ELECTRONIC FUNDAMENTALS SERIES

The manuals on electronic fundamentals form a progressive series of educational texts which present the theory and application of electronics for the military services. The series starts with the basic electrical fundamentals and extends to the most recent technical concepts, as applied to telegraphy, telephony, radio, loran, facsimile, radio direction finding, radar, meteorological radio, television, and other military equipments. See SR 310–20–4 for published available manuals in this series.

TECHNICAL MANUAL
No. 11-679

DEPARTMENT OF THE ARMY
WASHINGTON 25, D. C., *7 December 1953*

FUNDAMENTALS OF CARRIER AND REPEATER

		Paragraph	Page
CHAPTER 1.	INTRODUCTION	1–7	3
2.	TRANSMISSION LINES		
Section I.	Transmission line theory	8–22	8
II.	Transmission line practice	23–37	26
CHAPTER 3.	VOICE-FREQUENCY REPEATERS	38–70	43
4.	CARRIER TELEPHONY	71–117	84
5.	CARRIER TELEGRAPHY		
Section I.	Introduction to carrier telegraphy	118–124	125
II.	Telegraph circuits and transmission principles	125–138	130
III.	Loop circuits of carrier telegraph systems	139–149	145
IV.	Elements of carrier telegraph systems	150–167	168
V.	Characteristics of carrier telegraph systems	168–181	190
CHAPTER 6.	TRANSMISSION MEASUREMENTS		
Section I.	Transmission units	182–190	204
II.	Methods of transmission measurement	191–211	214
APPENDIX	LOGARITHMS AND DECIBELS		241
INDEX			248

*This manual supersedes so much of TM 11-475, 28 October 1944, as pertains to fundamentals of carrier and repeater.

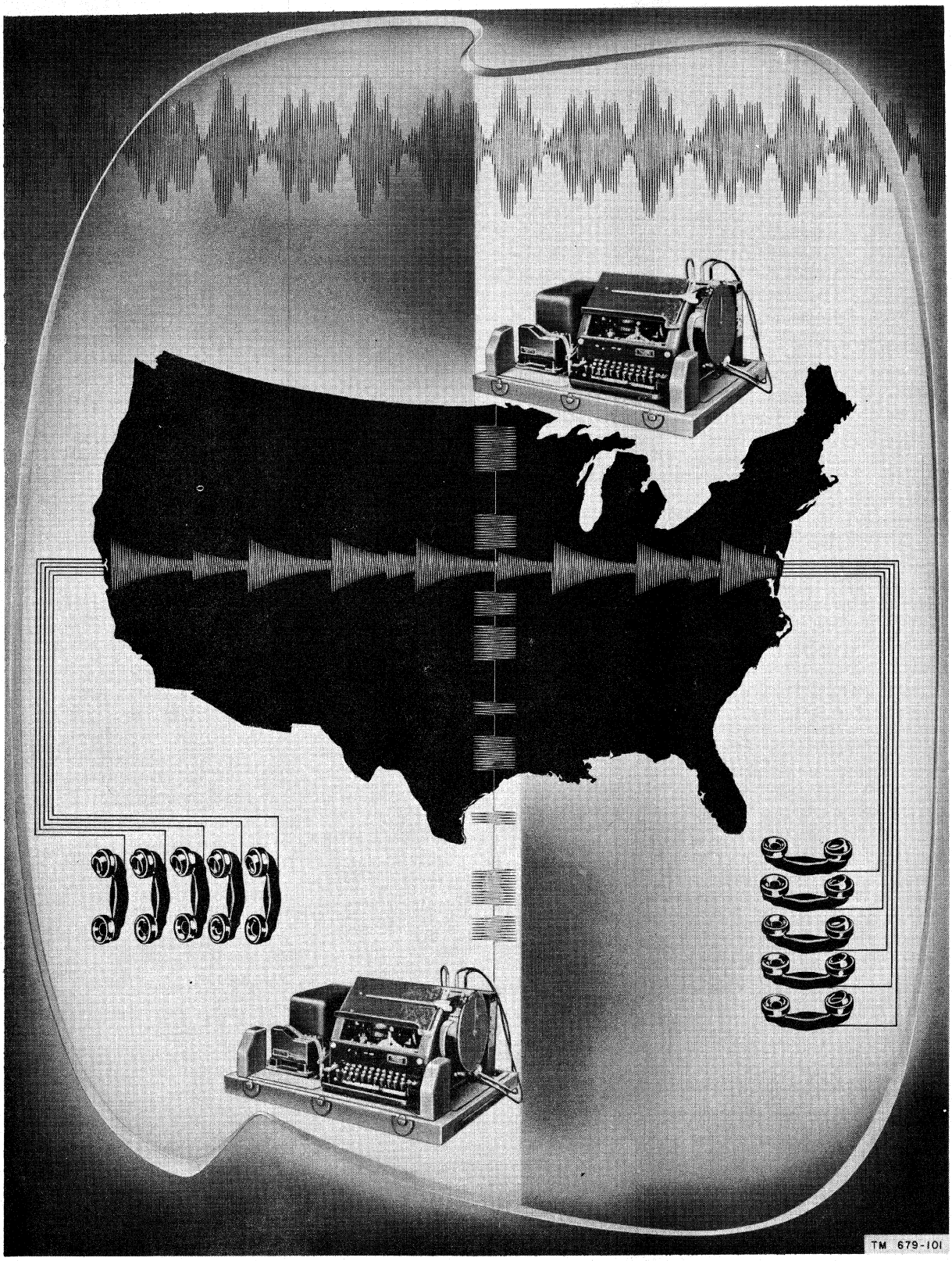


Figure 1. Nation spanned by carrier telephone and television systems.

CHAPTER 1
INTRODUCTION

1. Purpose of Repeater and Carrier

a. TM 11–678 discusses the subject of telephony and some of the older as well as present-day telephone systems. Although these systems have both civilian and military applications, their usefulness is limited. One of the limitations is the relatively small distance over which conversations can be transmitted effectively. Another limitation is that only one conversation can be transmitted over a single line at any one time.

b. With the rapid growth of the United States in the early part of this century, greater demands were made on telephone service. Not only were more telephones required to accommodate the larger population, but long-distance telephone service became more and more necessary. At the same time, the growth of the military services and their expansion into foreign areas made similar demands.

c. To keep up with these increasing demands, it was necessary to solve two major groups of problems: (1) how to transmit messages effectively over great distances, and (2) how to increase the number of telephone circuits without adding thousands of miles of wire. The purpose of this manual is to discuss the problems involved and their solutions by means of the various repeater arrangements and carrier systems in current use. Figure 1 shows symbolically our Nation spanned by carrier telephone and telegraph systems. This chapter defines some of the fundamental terms used in this manual, and introduces briefly the major principles and components of the carrier and repeater.

2. Repeater

a. Definition of Repeater. In telephony, the term *repeater* is used to designate a device for amplifying a signal that has been reduced in amplitude because of losses occurring in the process of transmission over wires. The principal component of a repeater is an electronic amplifier, usually a thermionic vacuum tube, which supplies new energy to the weakened signals. A repeater also includes various components such as repeating coils, equalizers, and hybrid sets, necessary to practical operation.

b. Use of Repeaters.

(1) Repeaters consist essentially of vacuum-tube amplifiers and their associated circuits. Amplifiers, however, are basically one-way devices; that is, they amplify in one direction only. Since a telephone system must provide for two-way communication, two amplifier circuits are required in most repeater arrangements. This is indicated in A, figure 2, where each amplifier is represented by a triangular symbol whose point indicates the direction of transmission. Each amplifier requires a pair of wires for connection to its input (grid) circuit, and another pair for connection to its output (plate) circuit; therefore, for *two-way* communication there is on each side of the amplifiers a total of *four* wires, two input and two output.

(2) A switchboard line circuit is a *two-terminal* device, as explained in TM 11–678. The *four* wires from the amplifiers in one side of the repeater must be coupled to the *two-line* circuit terminals of the switchboard jack (or to the two wires at the telephone set) without producing distortion. They may also have to be connected in intermediate locations. This problem is solved by the use of *hybrid arrangements* (B, fig. 2).

(3) The use of repeaters introduces problems. Among them is the possibility of introducing distortion of the signal because of interaction between adjacent amplifiers

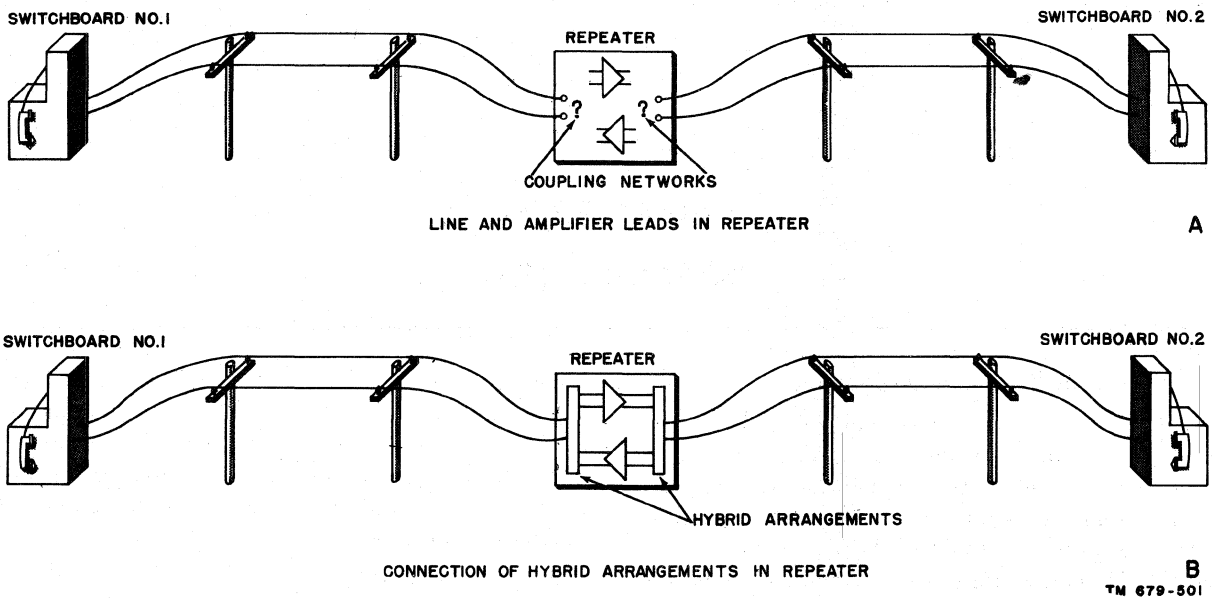


Figure 2. Connections of hybrid arrangement in basic two-way repeater.

and the response characteristics of the amplifiers. Certain devices, such as pads and equalizers, along with the correct spacing of the amplifiers, are effective in overcoming some of these problems.

(4) Basically, there are three types of telephone repeaters. They may be classified as the *21-type repeater*, which is a two-wire, two-way, one-amplifier device; the *22-type repeater*, which is a two-wire, two-way, two-amplifier device; and the *44-type repeater*, which is for use with four-wire systems.

3. Hybrid Arrangements

a. A hybrid arrangement is a device that permits the use of one-way amplifiers in two-way transmission circuits. It must accomplish this task without appreciable distortion. A, figure 2, shows the 2 wires from the line and the 4 wires from each side of the amplifiers. Although they must be connected, it is not feasible to connect the 4 wires to the 2 wires directly. If this were done, the amplified signal from one amplifier would pass through the other amplifier and arrive back at its input, where it would be amplified again. This would cause the circuit to oscillate or *sing*.

b. Some form of coupling network is called for which connects the two-wire line to the four wires from the amplifiers. The required coupling network, or hybrid arrangement, is indicated in B, figure 2. How and why the various hybrid arrangements work is considered in chapter 3.

4. Carrier System

a. Definitions.

(1) *Carrier telephony*, as used in this manual, refers to a system of telephone communication by which several conversations or messages may be transmitted simultaneously over a single circuit. This is accomplished by shifting the voice band of frequencies to other positions in the frequency spectrum.

(2) The term *carrier* refers to an oscillator-created a-c (alternating-current) wave, which is modulated by the message-bearing band of frequencies, and thus *carries* the intelligence. A system that uses a carrier wave is known as a *carrier system*. In present-day carrier systems, the carrier is actually suppressed, and one of the carrier sidebands, containing the original intelligence, is transmitted.

b. Basic Principles of Carrier System. Figure 3 illustrates the basic principles of the carrier system. The important steps in carrier telephony (left to right) are outlined in (1) through (7) below.

(1) *Generation of carrier wave* takes place in the vacuum-tube oscillator, in the figure called *carrier oscillator*.

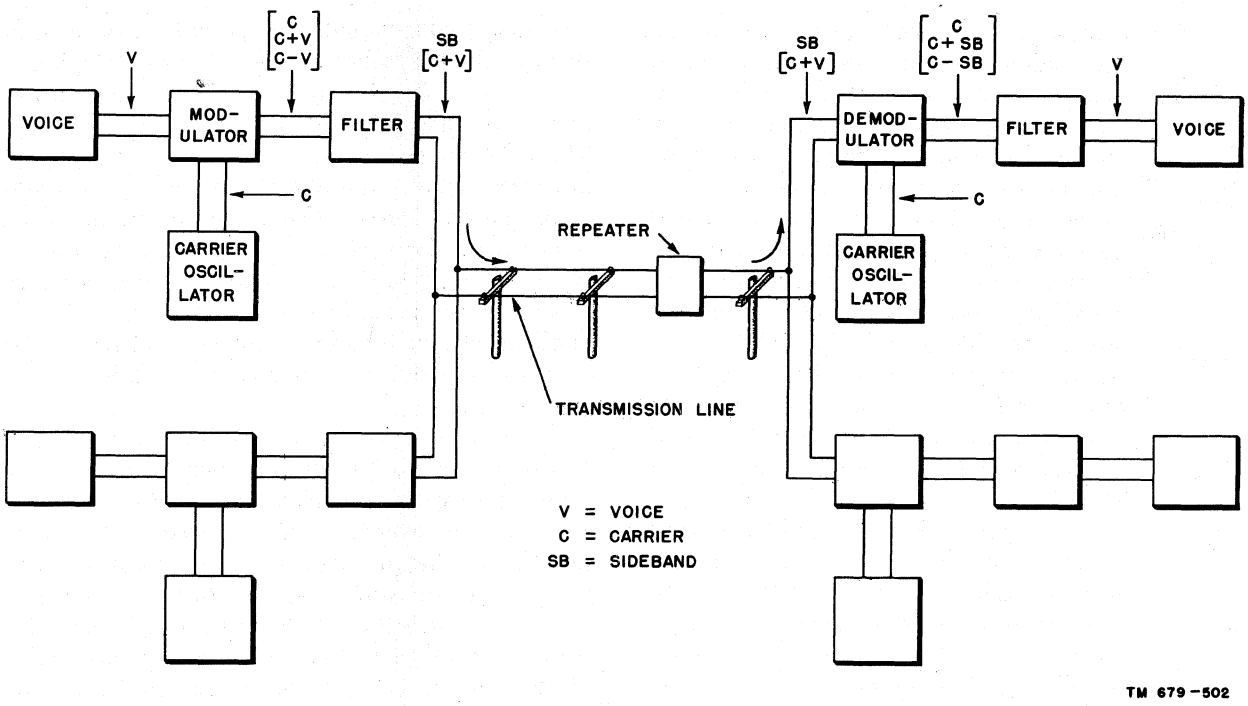


Figure 3. Two-channel one-way simple carrier system.

(2) *Modulation* means a change. In connection with carrier, modulation refers to the *change* which occurs in the carrier wave when it is combined with the voice frequencies in the *modulator circuit.* For example, the change in amplitude of the carrier is known as *amplitude modulation.* The carrier frequency, C, and voice frequencies, V, are introduced into different parts of the modulator circuit (fig. 3) The output, resulting from the action which takes place, contains the carrier frequency and two bands of new frequencies, called *sidebands.* The upper sideband, C plus V, contains frequencies which are the *sum* of the carrier and the various voice frequencies; the lower sideband, C minus V, contains frequencies which are the *difference* between the carrier and the voice frequencies. For example, if the oscillator produces a carrier of 25,000 cycles and the voice frequencies are 500 and 1,000 cycles, the output of the modulator would contain these frequencies: 25,500 and 26,000 cycles (the upper-sideband frequencies), 25,000 cycles (the carrier), and 24,500 and 24,000 cycles (the lower-sideband frequencies).

(3) There are other types of modulation besides the amplitude modulation in (2) above. Two common types are *frequency modulation,* in which the frequency of the carrier wave is caused to vary in step with the intelligence-bearing signal, and *phase modulation,* in which the phase of the carrier wave is caused to change in accordance with the intelligence-bearing signal. Of these additional two, this manual is concerned only with frequency modulation. Frequency shift technique, which is basically frequency modulation, is treated extensively in chapter 5. Other forms of modulation, particularly pulse-time modulation and pulse-code-modulation may be used in carrier communication systems in the future.

(4) Although the output of the modulator circuit contains the carrier, present-day systems do not transmit the carrier along with the sidebands. This requires means for removing or *suppressing the carrier.* Some older systems used selective *filter networks* to accomplish this, but more recently the modulator is designed so that it automatically eliminates the carrier from the output.

Filters are still used to remove one of the sidebands, however; consequently, only one of the sidebands (either the upper or the lower) is transmitted to the distant station. In figure 3, the upper sideband is being transmitted.

(5) Since electrical energy transmitted over great distances undergoes considerable attenuation, the electrical waves must be *amplified* at regular intervals to make up for the losses. This *amplification* is accomplished by means of repeaters.

(6) *Demodulation* is the reverse of modulation, described in (2) above. The circuit in which demodulation takes place is called the *demodulator* (fig. 3). The carrier, which is reinserted at the demodulator, combines with the sideband entering the demodulator in such a way as to reproduce the original voice frequencies. Other undesired frequencies are produced at the same time, however, so that the voice frequencies must be separated by means of suitable filters. The processes of modulation and demodulation are accomplished by vacuum tubes or (more commonly) by copper-oxide devices known as *varistors*.

(7) Although only the top half of figure 3, which illustrates one channel, is referred to in (1) through (6) above, it is possible to operate such a system with several carriers. This method is called *multichannel operation*. The lower half of the figure, for example, shows the addition of a second channel using a carrier frequency different from the first. Both channels can be operated at the same time, thus permitting the sending of two messages, each occupying a different portion of the frequency spectrum, simultaneously over the same line. The two messages are separated at the receiving end, each receiving channel accepting only the frequencies carrying one message. In a similar manner, many more channels can be added and operated over the same line. This is the advantage the carrier system offers.

c. Comparison of Carrier Telephony with Radio. Carrier telephony resembles radio broadcasting, the essential difference lying in the range of the carrier frequencies used. Radio uses carrier frequencies high enough to permit transmission through the air without wires. Carrier telephony, as discussed in this manual, uses much lower carrier frequencies, suited to transmission lines. There is another system of carrier telephony, however, called *radiotelephony*, where wireless transmission is used. This type of transmission is necessary for such communications as ship-to-shore and ship-to-ship messages, for which transmission lines cannot be erected. In some military applications, such as in combined land and water operations, carrier telephony is combined with radio links.

d. Problems of Carrier System. Any system which makes use of circuits such as those mentioned presents problems which must be carefully studied and overcome if the system is to be practical. The problems involved in repeater and carrier include such considerations as the prevention of distortion of the voice frequencies; the prevention of cross-talk, or undesired interaction of the various current waves transmitted simultaneously over a transmission line; and the use of effective means of amplifying the currents to permit long-distance transmission. These problems must be understood thoroughly by a student of repeater and carrier theory.

5. Military Multichannel Carrier Systems

a. General. Tactical multichannel carrier systems were used first by the military services during World War II. The systems which were used as tactical equipments usually are referred to as the CF series. The CF systems gave excellent service all through the late conflict, and are at present standard Signal Corps equipment.

b. Telephone Terminal CF–1( ) and Repeater CF–3( ). This carrier system provides one voice-frequency and three carrier telephone channels in the frequency range of 200 cycles to 11.8 kc (kilocycles). Each channel is 200 to 2,800 cps (cycles per second) wide. Generally, a transmission line commonly known as spiral-four is utilized. With normal repeater spacing of 25 miles, 150-mile transmission range may be obtained using five CF–3 repeaters. Therefore, this system creates three additional telephone channels, making a total of four voice-frequency channels.

c. Telegraph Terminals CF–2( ). The CF–2 telegraph terminal provides four duplex 60-wpm (words per minute) teletype channels on a channel of the CF–1 telephone carrier. Thus, each telephone channel may be converted into four teletype

channels. With suitable arrangements, each telephone carrier channel can take as many as 12 telegraph channels.

d. Packaged Equipments. In addition to the CF series, various commercial equipments, repackaged to facilitate installation, are used by the Signal Corps for rear area fixed plant applications. The packaged equipments are military adaptations of the commercial C-type and H-type carriers and repeaters. The C-type carrier provides three telephone channels in addition to the normal voice-frequency channel. The H-type carrier makes available one carrier channel on top of the original voice-frequency channel. Both systems are designed for use on open-wire lines.

e. New Military Carrier Systems. Although the CF series carrier equipment gives good point-to-point service, it does have limitations. In particular, there are the special problems of interoperation of carrier systems and the integration of carrier systems with radio circuits. These, coupled with the ever-increasing need for flexible communications, led to the development of the integrated communication system plan. This called for the development of 4-, 12-, and 48-channel carrier systems using cables or radio relay sets as transmission media. A brief description of the new four-channel system is given in *f* and *g* below.

f. Telephone Terminal AN/TCC–3( ). This carrier system provides four carrier derived channels in addition to the normal voice-frequency channel. The carrier derived channels provide a bandwidth of 300 to 3,500 cycles, assuring high quality message service. Each carrier channel, if desired, can provide eight duplex 100-wpm voice-frequency telegraph channels. Moreover, the bandwidth of the channels is such that they can be used for data transmission or facsimile. The frequency range of the system is 300 cycles to 20 kc, and, because of this high frequency limit, a special new lightweight spiral-four transmission line has been developed for use with the system.

g. Building Block Principle. With the realization of the importance that modern multichannel systems will play in future military planning, the design engineers have been faced with the problem of expanding these facilities. The new AN/TCC–3( ) is to serve as the basic four-channel building block for 4-, 12-, and 48-channel systems. Carrier systems using the building block principle one day will be in the field and, undoubtedly will give the American Army the finest, most rapid, and flexible military communications system in the world.

6. Summary

a. Repeater and carrier development was an outgrowth of the increase in population in the United States, and the resulting increased demands, both civilian and military, on long-distance telephone communications.

b. The major problems involved in this development resulted from the necessity for providing means for amplifying signals which are attenuated in the process of long-distance transmission.

c. These problems are overcome by the use of repeaters, which include amplifiers, and of hybrid arrangements, which permit effective coupling of the amplifiers to the line or to terminal stations.

d. Carrier systems permit the simultaneous transmission of several conversations over a single transmission line. In order to accomplish this, two important processes, modulation and demodulation, are involved.

e. This manual is organized to present the material in a logical sequence. The study of repeater and carrier will be facilitated if the material in each chapter is covered thoroughly before proceeding to the next chapter. Frequent review of principles previously learned will be an invaluable aid to mastery of the subject.

7. Review Questions

a. What is a repeater? What components may a repeater include?

b. What is the function of a hybrid arrangement in a telephone system?

c. Define: carrier system; carrier telephony.

d. Define: modulation; demodulation.

e. What is meant by the following terms: upper sideband? lower sideband?

f. Draw a labeled block diagram of a simple carrier system.

g. What are some of the important military carrier systems?

h. Discuss briefly some of the problems connected with carrier telephony.

i. What is meant by the building-block principle?

CHAPTER 2

TRANSMISSION LINES

Section I. TRANSMISSION-LINE THEORY

8. Function of Transmission Line

a. A transmission line provides a means for transferring electrical *power* from one point to another (fig. 4). In an electrical power distribution system, shown in B, the amount of power generated by the central powerhouse, and transferred over the line, is large. In a communication system, shown in A, the magnitude of the signal power transferred is small. In either system, the transmission line is a means of transferring power between two points.

b. Any communication or power distribution system has three essential components—a device to generate electrical power, such as a telephone transmitter or a powerhouse generator, located at the sending end of the transmission line; the transmission line; and a device that converts electrical power to some other useful form of power (such as a telephone receiver which converts electrical power to sound), located at the receiving end of the line. The transmission line system must always consist of 2 separate conductors, 1 of which may be the earth itself.

9. Distributed Constants

a. Line Characteristics. When a generator is connected directly to a power-consuming load, the load may be represented as a simple resistor, as in figure 5. This is because only the resistance of the load is capable of dissipating power. The power consumed in the resistor is equal to the power supplied by the generator. Assume that the same power-consuming device is connected to the generator by means of a long transmission line, as in figure 6. Although this circuit might appear identical with the directly connected circuit of figure 5, it is now found that in the second circuit the power consumed in the load is

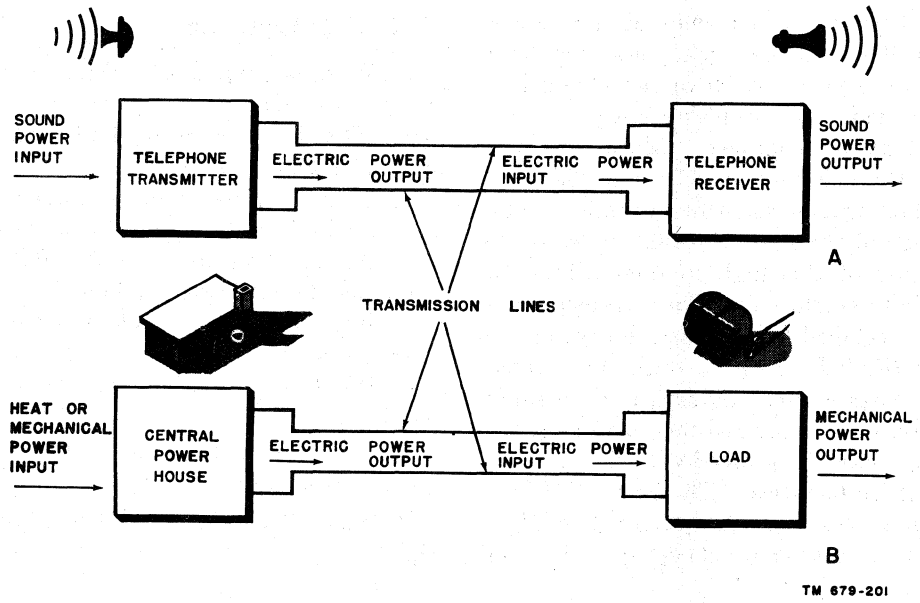


Figure 4. Two systems using transmission lines.

less than the power supplied by the generator. Thus, the transmission line has characteristics that cause a power dissipation (loss) between the generator and the load. This power dissipation does not take place at any one point, but occurs equally along the entire length of the line. The electrical characteristics of the line that *cause* the power loss are distributed uniformly over the entire length of the line. For this reason, the characteristics are called the *distributed constants* of the line. If the power dissipation took place at distinct points on the line because these characteristics were concentrated, as a coil concentrates inductance or a capacitor concentrates capacitance, they would be called *lumped constants*. There are four distributed constants—series resistance per unit length; series inductance per unit length; shunt capacitance per unit length; and shunt conductance (more commonly known as leakage) per unit length.

b. Series Resistance per Unit Length. The total resistance of a transmission line is found by multiplying its *series resistance* per unit length by the total length of line. Thus, if the series resistance of a line is 10 ohms per mile, and the length is 100 miles, its total resistance is 1,000 ohms. The distributed resistance of a transmission line depends on the size of the wires and the frequency of the traveling wave. The variation of resistance with frequency results from the phenomenon called skin effect. The resistance increases as the diameter of the wire decreases, and as the frequency of the transmitted wave increases. It is given the symbol r, and is expressed in *ohms per loop mile*. A loop mile (fig. 7) is the total length of both lines between two points 1 mile apart. Thus, a loop mile of line is 2 miles long, and the resistance in ohms per loop mile is twice the resistance of one conductor per mile. The series resistance per loop mile may be represented in an equivalent circuit (A, fig. 8), where half of it is considered to be lumped in series with each wire; or it may be as represented in B, where it is concentrated in one conductor only.

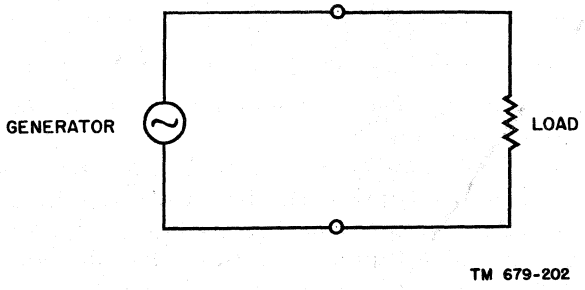


Figure 5. Generator and load connected directly.

c. Series Inductance per Unit Length. Self-inductance is that property of a circuit which causes a countervoltage to be induced in the circuit by a change in the current in the circuit. In a transmission line through which a changing current is flowing, a voltage is induced along the line. This indicates that *series inductance* is distributed over the entire length of the line. The magnitude of this series inductance is determined by the size of the wires and their separation. It increases as the center-to-center distance between the wire increases and as the diameter of the wires decreases. The distributed inductance is given the symbol l and is expressed in *henrys per loop mile*. The total inductance per loop mile may be represented also in two halves, each half representing the inductance in 1 of the 2 conductors (A, fig. 9), or it may be considered as concentrated in 1 conductor, as in B. The series inductance of a transmission line causes an opposition to the alternating voice currents in the form of inductive reactance. Inductive reactance, X_L, is a function of frequency and is expressed by the formula $X_L=2\pi fL$; therefore, as the frequency increases, the inductive reactance, or opposing effect, increases.

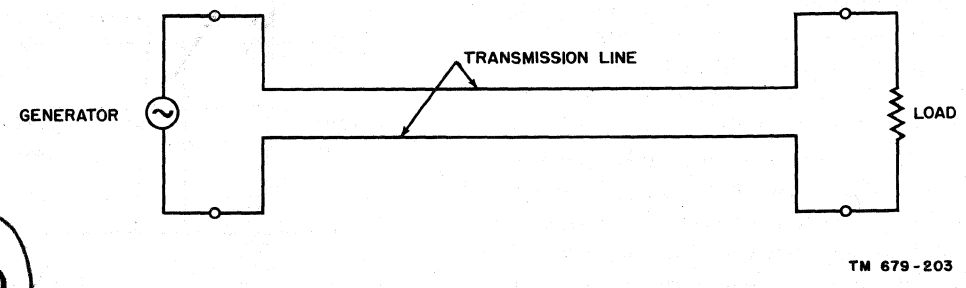


Figure 6. Generator and load connected by long transmission line.

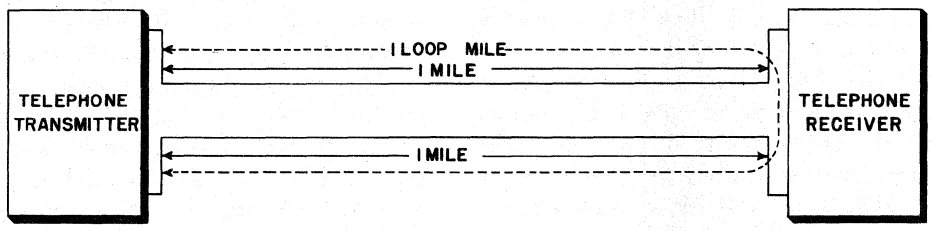


Figure 7. One loop mile of transmission line.

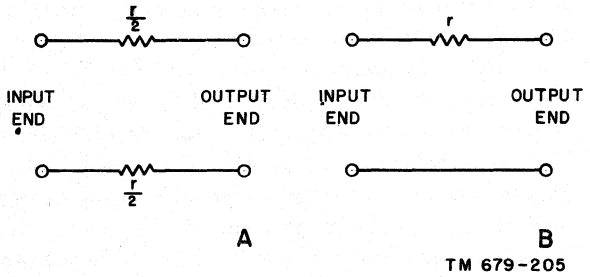


Figure 8. Distributed series resistance.

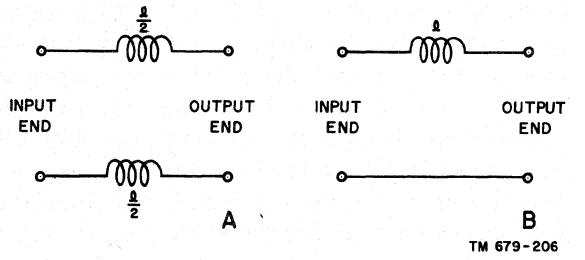


Figure 9. Distributed series inductance.

d. Shunt Capacitance per Unit Length. A capacitor consists of two metallic conductors separated by a nonconducting substance, such as air or some other *dielectric*. The capacitance is large when the area of the conductors is large, as in the case of two flat plates, and the capacitance increases when the distance between the plates is made smaller. In a capacitor, the separation of the plates deliberately is made a small fraction of an inch. A transmission line also consists of two metallic conductors separated by a dielectric. As a result, the line capacitance is distributed over the entire length of the line. However, the area of the surface of a length of transmission line is much less than the area of a conventional capacitor plate and the distance between the lines is much greater than the separation between the plates of a capacitor, so that the distributed capacitance of a transmission line is much less than the lumped capacitance of a capacitor. In the transmission line, the distributed capacitance appears between the adjacent wires and is called the *shunt capacitance*. The shunt capacitance per unit length is determined by the size of the wires, the distance between the wires, and the nature of the dielectric material between them. The capacitance increases as the diameter of the wires increases, as their center-to-center separation decreases, and as the dielectric constant increases. The distributed capacitance is given the symbol c; it is expressed in *farads per loop mile* and is represented as a capacitor shunted across the two conductors (fig. 10). The important thing to keep in mind is that the distributed capacitance causes capacitive reactance to develop across or in shunt with the transmission line. This causes a shunting of voice-frequency currents across the line, and the result of this is that less voice-frequency energy reaches the receiving end. Since capacitive reactance decreases as the frequency increases $\left(X_c=\frac{1}{2\pi fc}\right)$, the shunting effect becomes greater as the frequency increases.

e. Shunt Conductance per Unit Length. Because the dielectric between the two wires of a transmission line is not a perfect insulator, a leakage current exists between the two. The dielectric separates the wires of the transmission line over its entire length so that leakage exists at every

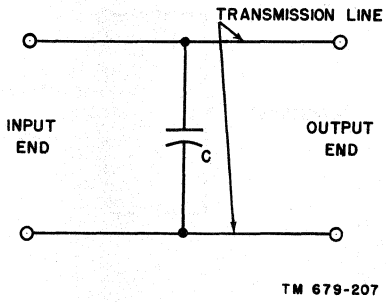


Figure 10. Distributed shunt capacitance.

point along the line. In open-wire line (par. 24) the dielectric between the conductors is air. Although dry air is an almost perfect insulator, outdoor air is seldom dry, and its conductivity increases greatly in damp weather. In cables (par. 25) the dielectric consists of the insulation around individual conductors. The best of insulators conduct extremely small amounts of current. Although cable leakage is unaffected by dampness of the outdoor air (which is excluded by the outer covering of the cable), it does vary somewhat with temperature. Since the leakage takes place through a conducting path between the wires this corresponding line characteristic may be called *shunt conductance*, or *leakage*. It is given the symbol *g;* it is expressed in *ohms per loop mile* and is represented as a resistance between the two wires (fig. 11). Note that line

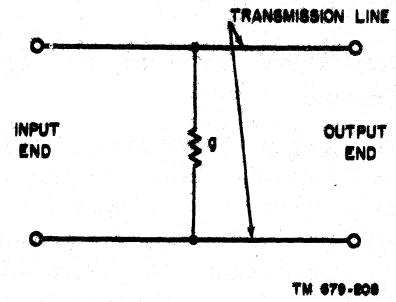


Figure 11. Distributed shunt conductance.

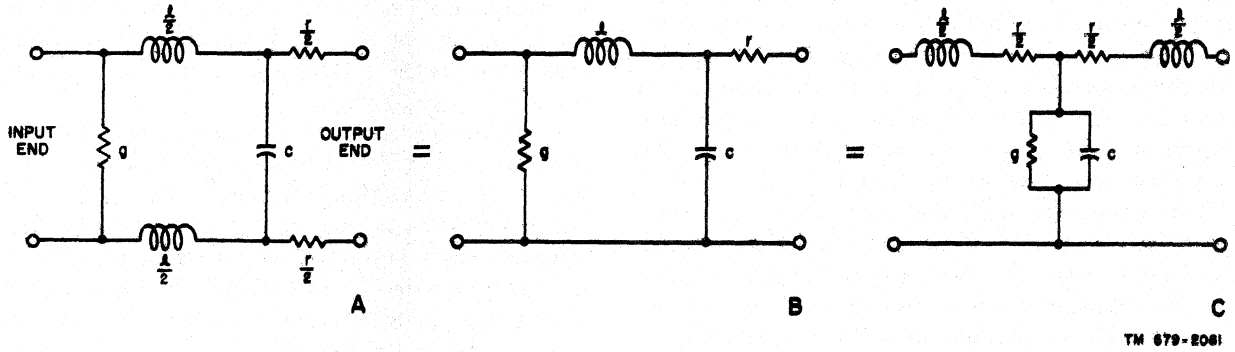
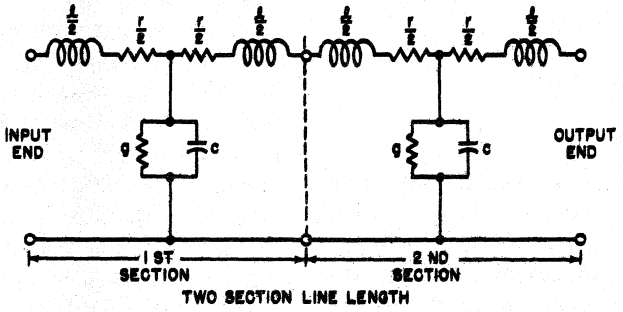


Figure 12. Line section of unit length.



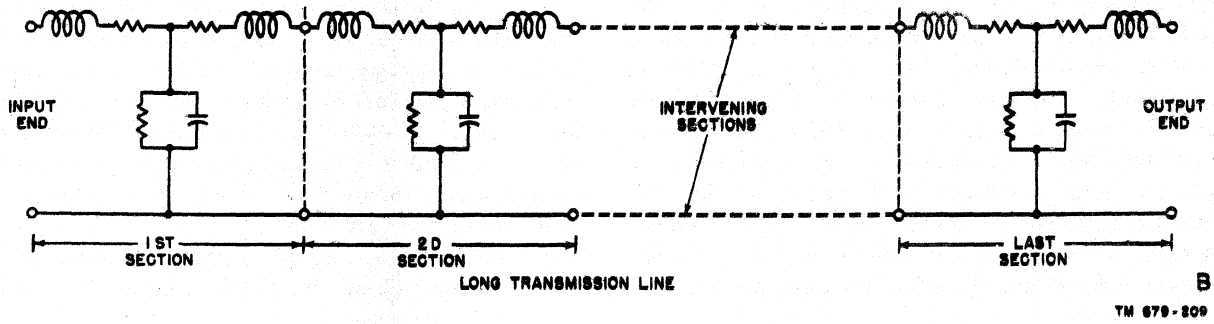


Figure 13. Equivalent network of transmission line.

leakage acts as a shunt to the flow of voice-frequency currents in the transmission line, and as a result, current shunted across the line cannot reach the receiving terminal.

10. Representation of Line by Equivalent Network

a. A unit length of transmission line, or line section, is 1 *loop-mile* of line. It may be represented for convenience as an equivalent network composed of all four distributed constants of the actual line, as in figure 12. Three alternative representations of the actual line section are shown. The equivalent tee section, in C, is formed by placing the series constants (resistance and inductance) in one line, half on each side of the shunt constants (capacitance and conductance). It must be remembered that in figure 12 the lumped elements, r, l, c, and g, are merely convenient representations of properties that are actually distributed over the whole length of the section. Their values are such that they would have substantially the same effect on a transmitted signal, and on any circuit connected to the line section, as the actual distributed constants of the section.

b. A transmission line more than 1 section long may be represented by an equivalent network formed by connecting 2 or more unit line sections, as in figure 13. The electrical behavior of the line then may be studied by analyzing the behavior of the equivalent network.

11. Radiation from Line

Under certain conditions, a transmission line acts like an antenna and radiates power. Such power, of course, is not delivered to the load. The input power to the transmission line, therefore, must be increased to make up for the power lost by radiation, just as it must be increased to make up for the power dissipated in the distributed line constants. At low frequencies, such as the frequencies used in normal telephone transmission, the amount of radiation is so small that it can be neglected. At higher frequencies, however, such as those used in radio transmission and some carrier telephone transmission, the radiation is greater, and continues to increase as the frequency increases. The radiation also increases with an increase in the length of the line. Radiation usually is not considered a line constant.

12. Characteristic Impedance of Line

The distributed constants of a transmission line determine its operating characteristics, which in turn affect the signal being transmitted over the line. The most important of its operating characteristics is the *characteristic impedance*, Z_o. The characteristic impedance of a transmission line is a property of the line itself, dependent only on its distributed constants. It is entirely independent of the length of the line, the internal impedance of the generator supplying the line, and the magnitude of the load placed across the terminals of the line. The characteristic impedance does, however, vary with the frequency of the transmitted signal. The importance of characteristic impedance lies in the requirements it imposes on the generator and load impedances to obtain maximum transfer of power to the load, as explained in paragraph 15.

a. To demonstrate the meaning of characteristic impedance, the tee section representation in A, figure 13, is transformed first to the pi section, composed of three resistances, of figure 14. Resistances alone are used, because the mathematics necessary to derive the value of characteristic impedance of an actual transmission line are beyond the scope of this manual. Using only resistances, the mathematics are much simpler, and the meaning of the results is the same as far as characteristic impedance is concerned.

b. Let the three resistances of the pi section (fig. 14) be 300 ohms each (A, fig. 15). The input impedance of this line section, then, is 200 ohms, as is demonstrated. Now, connect a second section, ahead of the first, making the line two sections long, as in B. The first section acts as an equivalent 200-ohm resistor across the second. Combining the resistances in the manner shown, it is found that the input impedance of a line two sections long is 175 ohms. Next, let the line be three sections long, as in C. The first two sections act as an equivalent 175-ohm resistor across the third section. The resistances are combined in the manner demonstrated, resulting in an input impedance of 173.3 ohms for a line three sections long. If the line were composed of 4 sections, the first 3 would act as an equivalent 173.3-ohm resistor across the fourth, and the input impedance then would be 173.2 ohms.

c. The values of input impedance found in *b* above are plotted on the graph in figure 16. The

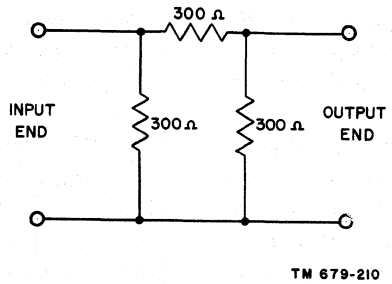
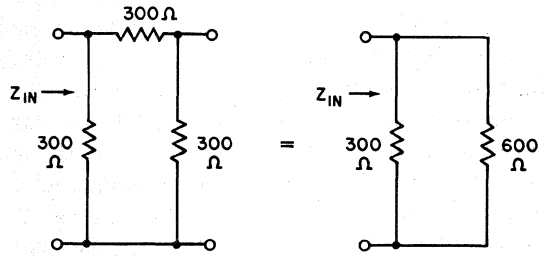


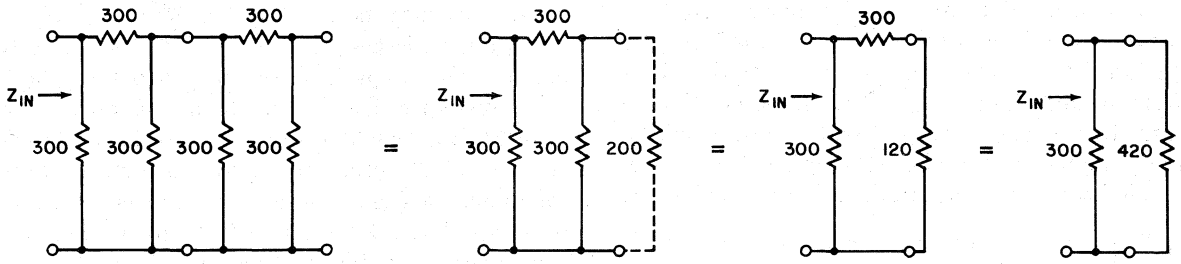
Figure 14. Unit line section represented as pi section.



$$Z_{IN} = \frac{300 \times 600}{300+600} = 200\,\Omega$$

INPUT IMPEDANCE OF ONE LINE SECTION

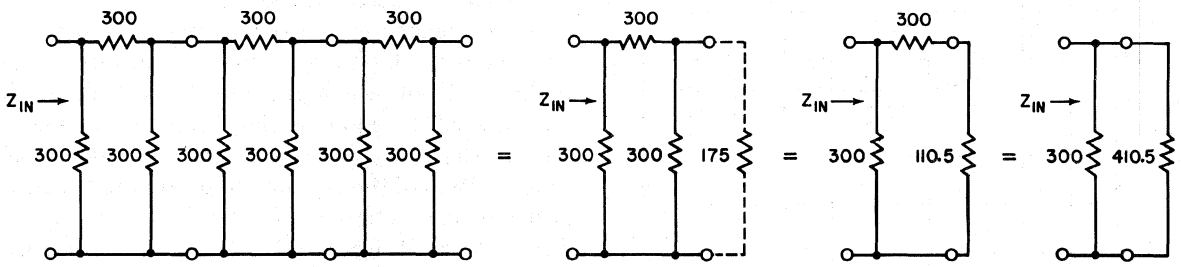
A



$$Z_{IN} = \frac{300 \times 420}{300+420} = 175\,\Omega$$

INPUT IMPEDANCE OF TWO LINE SECTIONS

B



$$Z_{IN} = \frac{300 \times 410.5}{300+410.5} = 173.3\,\Omega$$

INPUT IMPEDANCE OF THREE LINE SECTIONS

C

TM 679-211

Figure 15. Calculation of input impedance.

graph shows that the input impedance of the transmission line decreases toward a fixed value as the length of the line increases. Note that this is true because the amount of decrease becomes smaller and smaller as more and more sections are added. The input impedance cannot decrease to zero because of the distributed series resistance. When the line is very long, the addition of more sections causes practically no change in the input impedance; that is, the input impedance takes on a constant value. This constant input impedance of a very long line is called its *characteristic impedance*. Theoretically, the transmission line must be infinitely long in order that the input impedance be equal to the characteristic impedance of the line. But the graph shows that (for the example chosen) after taking only four sections, the line has an impedance value very nearly equal to its characteristic impedance.

d. For an actual line with distributed inductance and capacitance, the input impedance also approaches the characteristic impedance as additional sections are connected, but more slowly.

e. In the example in *d* above, the inductance and the capacity of the line were omitted for sake of simplicity. For an actual transmission line, however, their presence causes the characteristic impedance to vary with frequency. If the characteristic impedance for a transmission line is assumed to be 600 ohms at 1,000 cycles, it will be slightly higher at frequencies above 1,000 cycles, and slightly lower at frequencies below 1,000 cycles. In most cases, the differences in the value of the characteristic impedance are not great enough to be a source of difficulty.

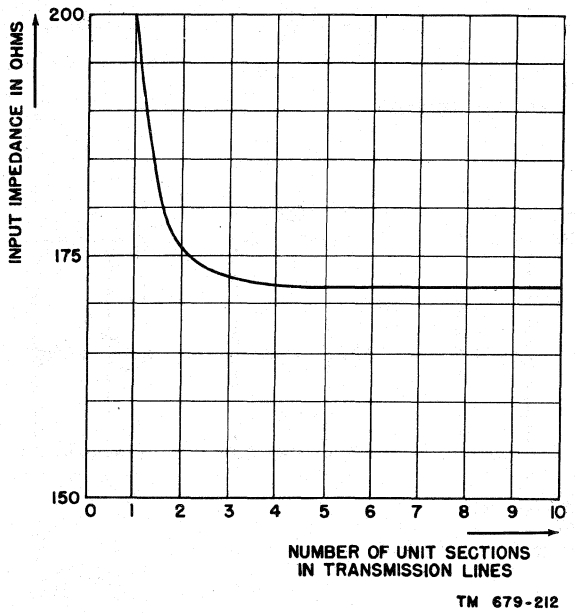


Figure 16. Input impedance versus transmission-line sections.

13. Terminating Impedances of Line

The input impedance of a transmission line is also equal to the characteristic impedance of the line when, no matter how short or how long, it is terminated by an impedance equal to its characteristic impedance.

a. To demonstrate this, a unit line section again will be represented as a pi section composed of three 300-ohm resistors (fig. 14). Again, this is not the true representation of a transmission-line section; however, the meaning of the results is practically the same as the meaning of the results achieved in the more difficult problem involving shunt capacitances and series inductances. Let the load across the line section be a variable resistor (A, fig. 17). The input impedance of the section is equal to the characteristic impedance of the line when the terminating impedance also is equal to the characteristic impedance of the line—in this case, 173.2 ohms.

b. When the load resistance is 50 ohms, the resistances may be combined as shown in B, and the input resistance is only a little less than 160 ohms. The graph in figure 18 shows how the input impedance varies as the load resistance ranges from 50 ohms and 300 ohms. When the load resistance is. 173.2 ohms, the input impedance may be calculated in the same manner, and is found also to be equal to 173.2 ohms. In figure 16, however, it is shown that the line section of figure 14 has a characteristic impedance of 173.2 ohms. The obvious conclusion is that when a line section is terminated by an impedance equal to its characteristic impedance, the input impedance is equal to the characteristic impedance.

c. Any length of the same type of line may be connected to the input of the section represented in figure 17, and the input impedance of the entire line still will be equal to the characteristic impedance of the line. In A, figure 19, the first section is one section of the new line connected to the input of the second section. The latter is the terminated line section of figure 17. Its input impedance already has been shown to be 173.2 ohms. This is the impedance in which the first, or new, section is terminated. The new section acts exactly as though it were terminated in a 173.2-ohm resistor, as shown in the equivalent circuit, at the right in A, figure 19. It has been shown, however, that a

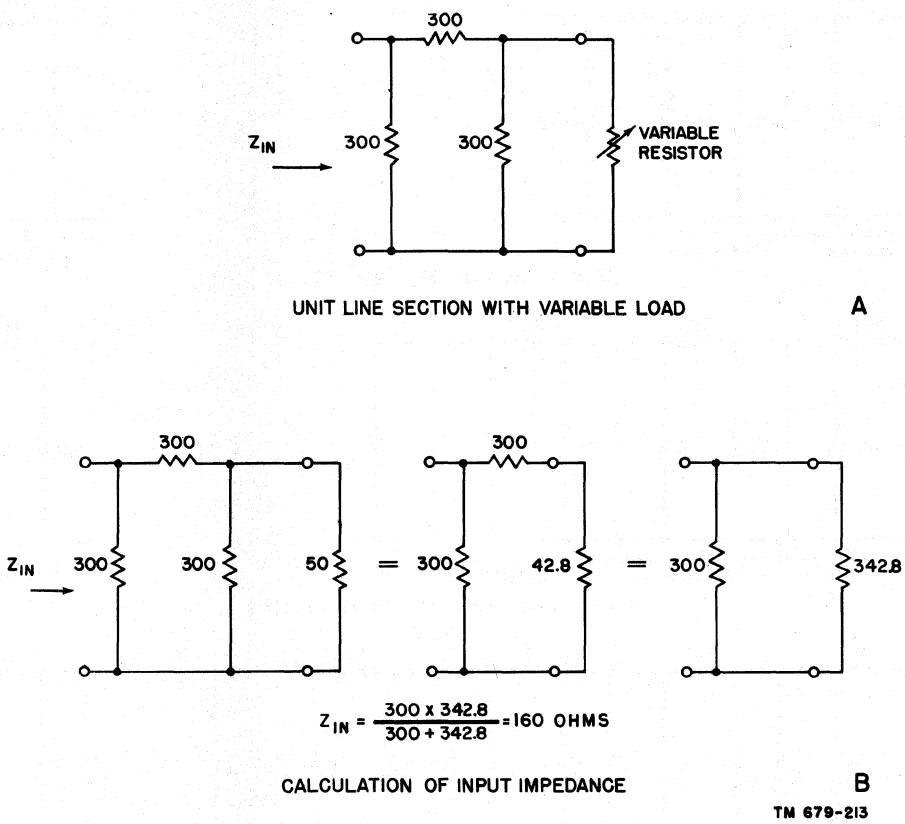


Figure 17. Variation of input impedance.

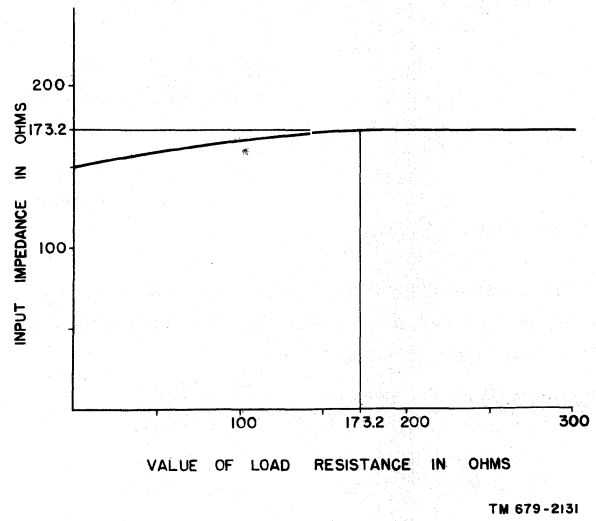


Figure 18. Input impedance versus load resistance.

section of line so terminated has an input impedance of 173.2 ohms. If another section of line is connected to the input of this terminated line—making a line three sections long, as in B—the same result is obtained. The newly added section is terminated in the input impedance of the old sections. Since this was equal to the characteristic impedance of the line, the new input impedance of the three sections is also equal to the characteristic impedance of the line. As the line increases in length, as in C, each newly added section is terminated in an impedance equal to its characteristic impedance; therefore, the input impedance of the entire line is also equal to the characteristic impedance of the line. Also, the impedance looking toward the load from any point on the line is equal to the characteristic impedance.

d. Paragraph 12*c* shows that the input impedance of a very long line is substantially equal to its characteristic impedance. This is true even when the long line is open-circuited or short-circuited at its load end (par. 12*b*).

e. The facts in *d* above may be summarized thus: The input impedance of a transmission line is equal to its characteristic impedance (1) when the line is very long, no matter what its termination, or (2) when it is terminated in a resistance equal to its characteristic impedance, no matter what its length.

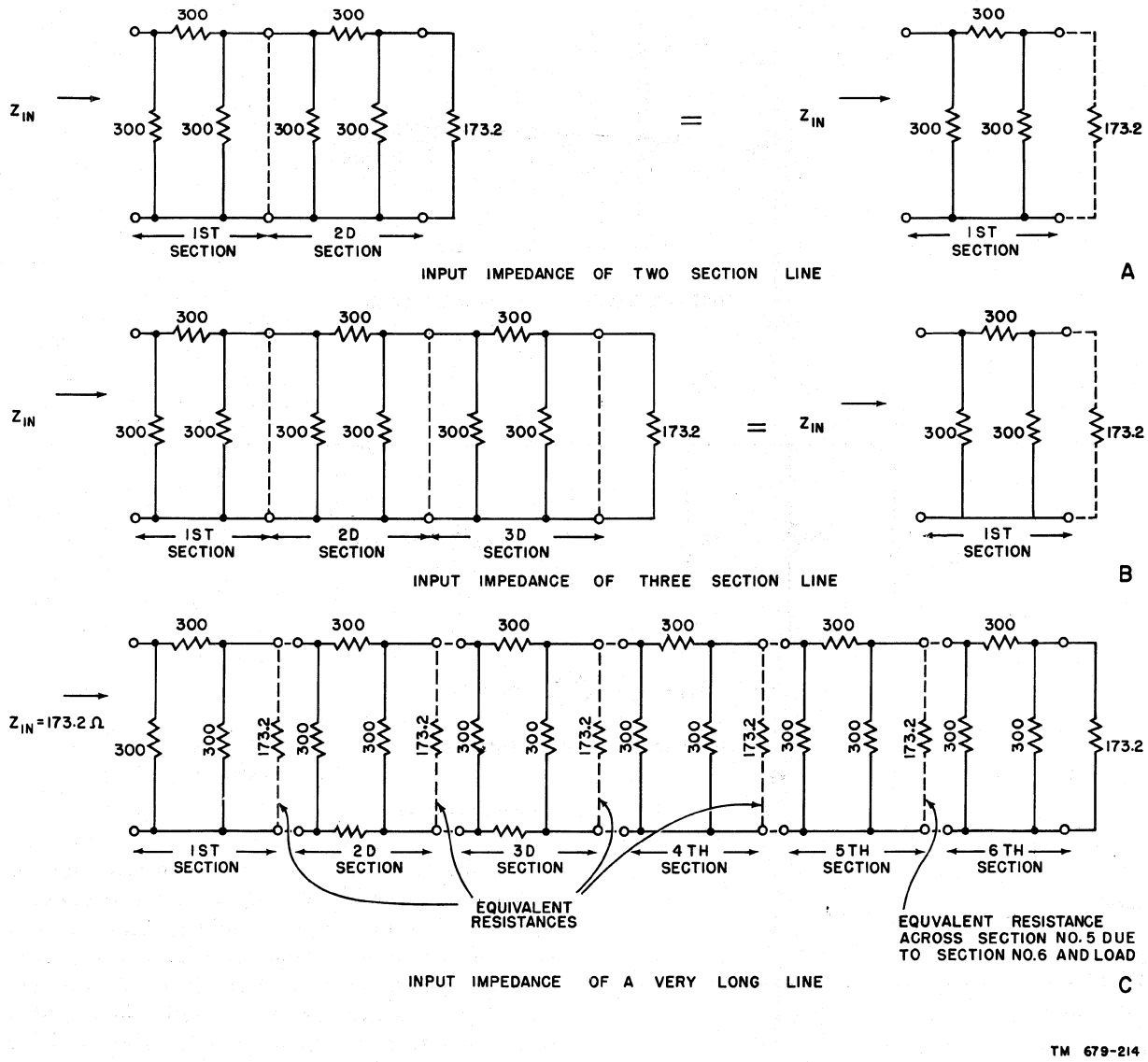


Figure 19. Input impedance of line terminated by Z_0.

14. Factors Affecting Characteristic Impedance

The value of the characteristic impedance of a transmission line is determined by the values of the distributed constants of the line. These are determined by the size of wire used, the distance between the wires, and the nature of the dielectric between the wires. The value of the characteristic impedance increases as the diameter of the wire decreases, as the center-to-center separation increases, and as the value of the dielectric constant decreases. Thus, the characteristic impedance is determined by the construction of the transmission line and is in no manner dependent on the length of the line. In addition, characteristic impedance varies with frequency. Typical values of the characteristic impedance for different types of lines, measured at 1,000 cps and with standard 8-inch spacing of wires, are—

104-gage copper wire	600 ohms,
128-gage copper wire	586 ohms,
165-gage copper wire	550 ohms,
Field Wire W–143	168.5 ohms.

15. Maximum Power Transfer

The primary purpose of a transmission line is to transfer a maximum amount of power to the receiver at the receiving end of the line. The *maximum power transfer* occurs when the input impedance of the receiver and the internal resistance of the generator are equal to the charac-

teristic impedance of the transmission line. For this reason, the value of characteristic impedance of a transmission line is important.

a. Consider the circuit in A, figure 20, which shows a generator delivering power to a 600-ohm resistor. Let the internal resistance of the generator be 600 ohms and let its voltage be 120 volts. The calculations show that the total output power of the generator is 12 watts, and that the power delivered to the 600-ohm load resistor is 6 watts. In B, the load resistor is changed to 300 ohms, causing the total output power of the generator to change to 16.0 watts, and the power to the load to change to 5.33 watts. In C, the load resistor is changed to 900 ohms, causing the total output power of the generator to change to 9.6 watts, and the power to the load to change to 5.76 watts.

b. The values of the power delivered to the load resistor calculated in *a* above, supplemented by others to make a smooth curve, are plotted in the graph of figure 21. The graph shows that maximum power is transferred from the generator to the load when the internal impedance of the generator and the impedance of the load are equal.

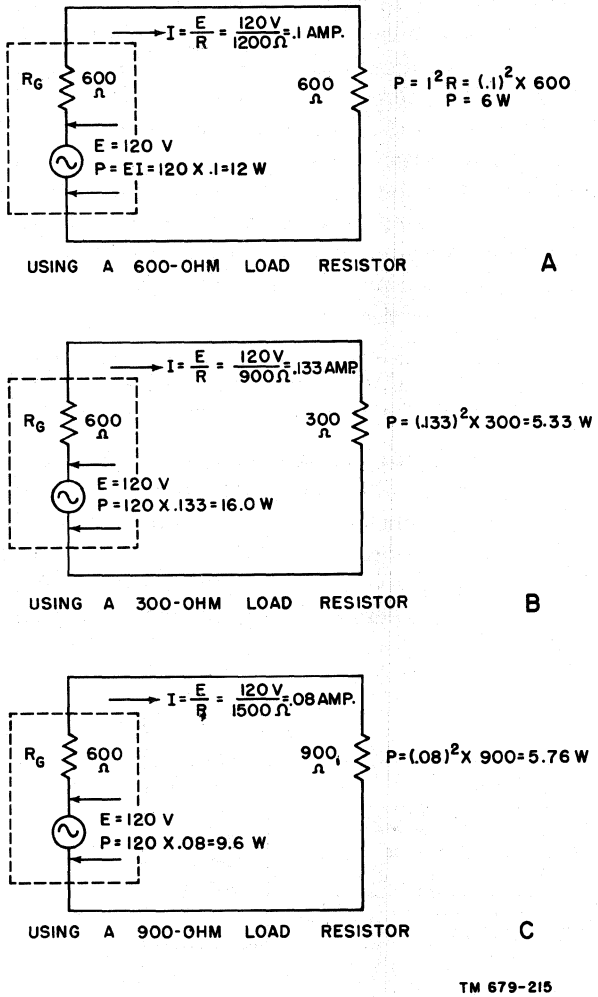

I = E/R = 120V/1200Ω = .1 AMP.
RG
600 Ω
600 Ω
P = I²R = (.1)² X 600
P = 6 W
E = 120 V
P = EI = 120 X .1 = 12 W
USING A 600-OHM LOAD RESISTOR
A
I = E/R = 120V/900Ω = .133 AMP.
RG
600 Ω
300 Ω
P = (.133)² X 300 = 5.33 W
E = 120 V
P = 120 X .133 = 16.0 W
USING A 300-OHM LOAD RESISTOR
B
I = E/R = 120V/1500Ω = .08 AMP.
RG
600 Ω
900 Ω
P = (.08)² X 900 = 5.76 W
E = 120 V
P = 120 X .08 = 9.6 W
USING A 900-OHM LOAD RESISTOR
C
TM 679-215

Figure 20. Generator operating into variable load.

c. The conclusion in *b* above is related to the characteristic impedance of a transmission line as follows: To the generator supplying power to a transmission line, the line is the load; consequently, the generator will deliver maximum power when its internal impedance is equal to the input impedance of the line. At the receiving end of the line, the line acts as the generator, delivering power to the load. The internal impedance of the line—here, the generator—is its characteristic impedance. Therefore, it will deliver maximum power to the load when the load impedance is equal to the characteristic impedance of the line. This being the case, the input impedance of the line is also equal to its characteristic impedance, requiring that the generator impedance also equal the characteristic impedance of the line. Thus, a 600-ohm line should be terminated by a 600-ohm load and supplied from a 600-ohm generator. Then maximum power transfer takes place at both ends of the line, and, therefore, from the actual generator to the actual load. Under this condition, the generator and the load are said to be *matched* to the line.

16. Attenuation in Line

A wave traveling along a transmission line is *attenuated* as it moves away from the generator.

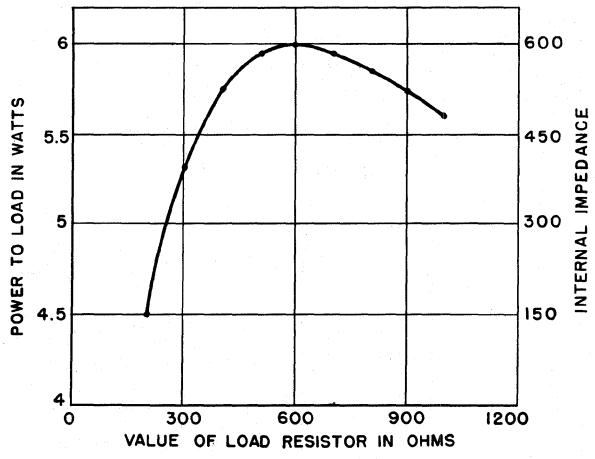


Figure 21. Power transferred to variable load.

The attenuation amounts to a gradual diminishing of the amplitude of the current and voltage waves.

a. Consider a transmission line which is terminated by a load having an impedance equal to its characteristic impedance (fig. 22). Because of the presence of the distributed series resistance of the line, there will be a series voltage drop (IR) along its entire length. In addition, some of the current will not travel over the entire line and reach the load, because of the distributed leakage and capacitance between the wires through which the current returns to the generator. The magnitude of both the current and the voltage waves, therefore, is diminished; and since the power in the line is the product of the current and voltage, the power gradually falls off as the distance from the generator lengthens. Figure 22 shows how the amplitude of the voltage, or the current, of the transmitted wave is decreased by the series resistance and the shunt leakage.

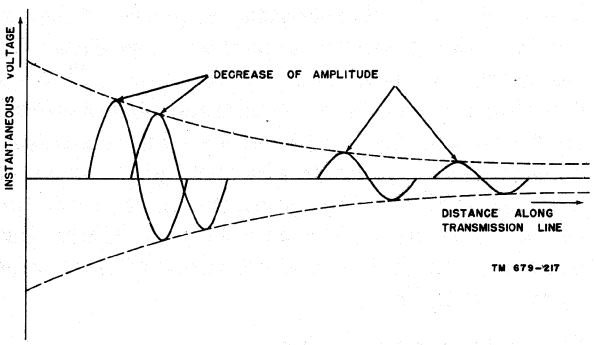


Figure 22. Attenuation of current or voltage along a transmission line.

b. If the distributed constants of the line are uniform over the entire length of the line—that is, if the distributed constants have the same values for every unit section—the amplitudes of the voltage and current waves decrease at the same rate over the entire length of the line. Figure 23 represents a line of at least four unit line sections. The impedance of the line is 600 ohms at all points on the line. The voltage at the input terminals of the line is 600 millivolts (fig. 23) and at the output terminals of the first section it is 60 millivolts. The decrease of voltage amplitude produced by the line section is in the ratio 10 to 1. (This ratio is selected as an example.) The decrease of current amplitude is in the same ratio. Thus, if the input current has a value of 1 milliampere, the output current will have the value of .1 milliampere. For the second line section, the input and output voltages are 60 millivolts and 6 millivolts respectively, and the input and output currents are .1 milliampere and .01 milliampere. The figures show that in the first line section the voltage amplitude is decreased by 540 millivolts, and in the second line section the voltage amplitude is decreased by 54 millivolts; but, in both, the ratio of input and output voltages is 10 to 1. In the current-wave amplitude, the amount of decrease is .9 milliampere in the first section, and .09 milliampere in the second section; but the ratio of input and output currents is 10 to 1 for either section. In other words, the *amount* of decrease differs from section to section but the *ratio* of input and output values of current and voltage is the same for all sections, as long as they have the same characteristics.

Note. The ratio of 10 to 1 used here is not the same for all types of lines, but varies widely according to the properties of the line as well as the length of the section.

c. In A, figure 24, the values of the voltage amplitudes at different points on the line are plotted and connected by a solid line. In B, the values of the current amplitudes are treated in the same way. Note how the current and voltage amplitudes decrease as the wave travels toward the

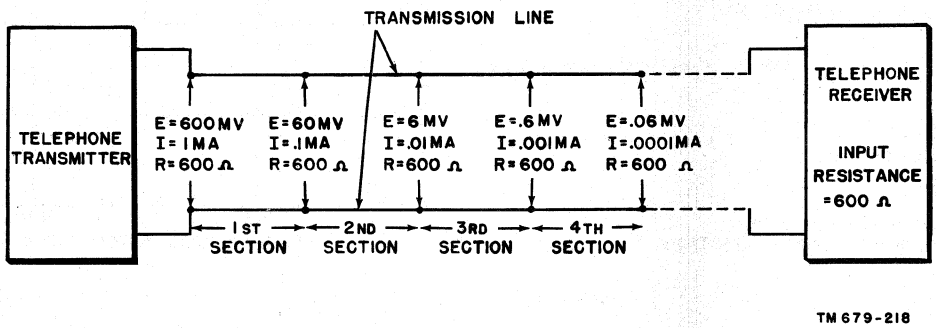


Figure 23. Attenuation of current and voltages.

load. The broken lines in both figures represent the values of current and voltage amplitude that would exist on a theoretically ideal line. An ideal line does not exist, because it is impossible to construct one without losses; the concept, however, serves an important purpose, because all practical cases may be referred to it for comparison. Also, under certain conditions, an actual line may approach the performance of a lossless (ideal) line. Such a line has no series resistance or shunt conductance; consequently, there is no power loss. As a result, the current and voltage amplitude will remain constant over the entire length of the line, as indicated by the straight broken lines in both A and B. Figure 24 shows, then, that if there were no series resistance or shunt conductance in the line, the transmitted wave would be of constant amplitude, but that the presence of series resistance and shunt conductance causes the amplitude of a transmitted wave to be attenuated at a constant rate.

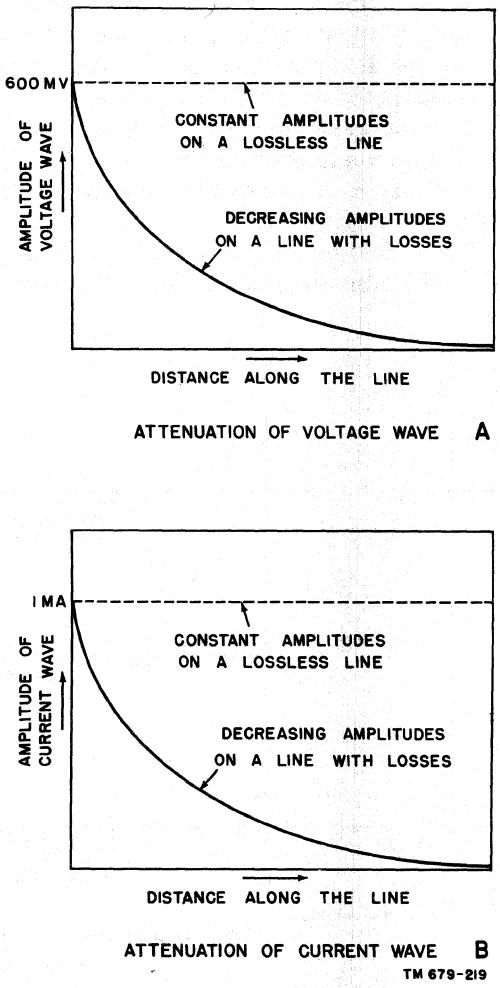

600 MV
AMPLITUDE OF VOLTAGE WAVE
CONSTANT AMPLITUDES ON A LOSSLESS LINE
DECREASING AMPLITUDES ON A LINE WITH LOSSES
DISTANCE ALONG THE LINE
ATTENUATION OF VOLTAGE WAVE A
1 MA
AMPLITUDE OF CURRENT WAVE
CONSTANT AMPLITUDES ON A LOSSLESS LINE
DECREASING AMPLITUDES ON A LINE WITH LOSSES
DISTANCE ALONG THE LINE
ATTENUATION OF CURRENT WAVE B
TM 679-219

Figure 24. Attenuation of transmitted wave.

17. Phase Shift in Line

The transmission line also causes a continuously increasing shift in the phase (*phase shift*) of the transmitted wave as it travels along the line toward the load.

a. Phase Velocity. When a wave is introduced on a transmission line at the sending end, a very small but definite amount of time goes by before it reaches the receiver. The wave moves along the line with a very high, but definite, velocity. Refer to the sine-wave signal shown in A, figure 25. Point X represents a definite phase of the wave—in this case, the positive maximum phase. X is not a fixed point, but an imaginary one that should be thought of as attached to the wave itself. Thus, X moves down the line as the electrical wave moves. The velocity of X as it moves along the line is the velocity of the total wave. This is called the *phase velocity*, because it is the velocity with which a point representing a phase of the wave moves forward. Further, it is the velocity with which a carrier signal moves along a transmission line. The phase velocity is not the same in all lines, but is determined by the characteristics of the line.

b. Phase-Shift Constant.

(1) Figure 25 illustrates how point X, the peak value of the wave, moves down the transmission line. It is important to remember that X and Y, in A, move with the wave, whereas points M and N, in B, are fixed on the line. The waveform shown in A is an instantaneous *snapshot* at a particular moment of time—that is, a graph of instantaneous voltage at a particular instant, plotted against distance along the line. A short time later, X will occupy the position of Y, whereas the instantaneous value of the signal at M passes through a complete cycle. Thus, the phase represented by point X will have progressed from M to N. Meanwhile, Y will have moved to a new position farther along the line; or, with respect to the transmission line shown in figure 25, it will have been absorbed in the receiver. This means that the wave has moved forward a distance exactly

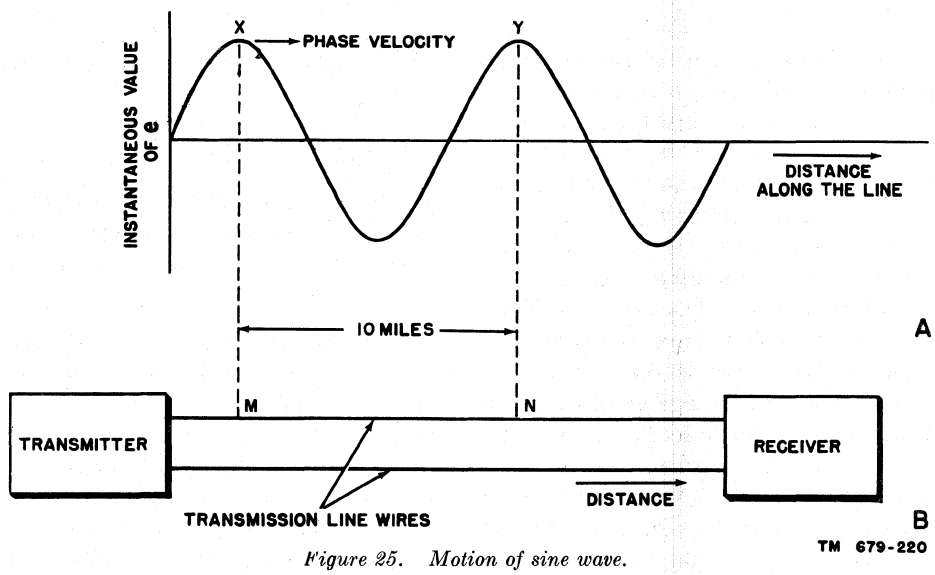


Figure 25. Motion of sine wave.

equal to the distance between two maximum points—that is, the distance between two points of similar phase.

(2) This distance is called the *wavelength* of the transmitted wave. The phase difference over this distance is 360°. If it is assumed, for example, that the wavelength is 10 miles, there is a phase shift of 360° over that length of line. In 1 mile, the total phase shift is 36°. The *phase-shift constant* (also called phase constant) is the phase shift in degrees per unit length of line. If 1 mile is considered as a unit length of line, the phase-shift constant in the preceding example is 36° per mile.

c. Phase Difference. In a line the distributed constants of which are uniform over its entire length, the phase-shift constant is also uniform. Furthermore, if the entire length of the line is not an *exact* multiple of the phase shift per unit length, there will be a *phase difference* between the sending and receiving ends. For example, if the phase-shift constant is 36° per mile on a uniform line 24 miles long, then the total phase shift is 24 times 36°, or 864°. Dividing this by 360° gives two complete cycles of the wave, with a remainder of 144°. The 144° represents the phase difference between the transmitted wave and the received wave.

18. Variation of Phase-Shift Constant

The phase-shift constant is determined by both the *frequency* and the *phase velocity* of the transmitted wave. How the phase-shift constant is affected by the frequency of the transmitted wave will be considered first. The line is considered to be ideal.

a. In A, figure 26, the sine-wave signal has a frequency of 10,000 cycles per second. M on the wave moves along the line to N, with a velocity that is characteristic of the properties of the line. In B, the sine-wave signal has a frequency of 20,000 cps. X moves along down the line to Y with the same velocity as M, since the velocity is fixed by the properties of the line.

b. When the frequency is 20,000 cycles, 2 cycles occur in the same time that 1 cycle takes place at a

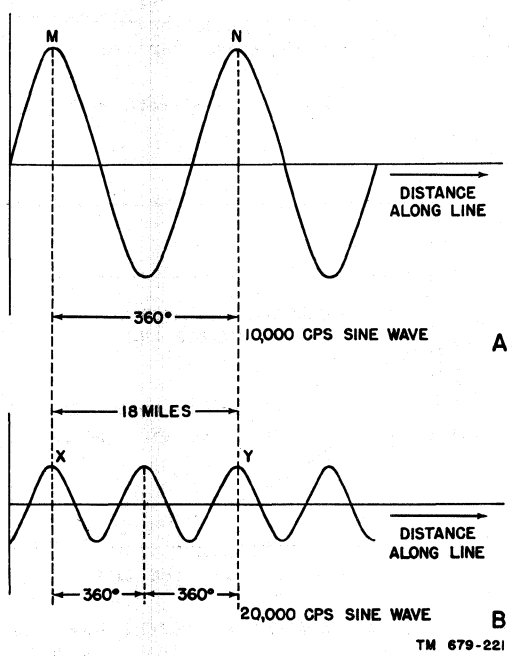


Figure 26. Motion of waves of different frequency.

frequency of 10,000 cycles. This means that during the time required for 1 cycle to move from M to N, in A, 2 cycles will move over the same distance, from X to Y, in B. Assume the distance M–N and X–Y to be 18 miles. In A, there is a phase-shift of 360° in 18 miles, and consequently the phase-shift constant is 20° per mile. In B, there is a phase shift of 720° in 18 miles; therefore, the phase-shift constant is 40° per mile.

c. These results show that when the frequency of the transmitted wave on an ideal line is doubled, the phase-shift constant is doubled also. If the frequency had been tripled, the phase-shift constant would have been tripled also. On the ideal line, then, the phase-shift constant varies directly with the frequency of the transmitted wave.

d. The phase-shift constant is determined also by the phase velocity of the transmitted wave. This can be shown easily by setting up and examining their relationship as follows:

Let V=the phase velocity in number of miles per second.

Let β=the phase-shift constant in number of degrees per mile.

Then (1) $V\beta$=the number of degrees per second, (2) $360° f$=the number of degrees per second.

For example, if $f=1$ we have 1 cycle per second and, therefore, 360° per second. There are 360° in each cycle.

If $f=2$, we have 2 cycles per second and, therefore, 360° times 2 equals 720° per second.

Equating (1) and (2), $V\beta=360° f$.

Transposing

$$(3)\beta=\frac{360° f}{V}$$

where β=the phase-shift constant,
V=the phase velocity,
f=the frequency.

e. This simple equation immediately shows the relationship between β, f, and V. V depends only on the distributed constants of the line; f is the frequency of the signal. For example, if f is a constant, and V increases, β decreases. Furthermore, since the phase velocity, V, is determined by the distributed constants of the line, the phase-shift constant, β, also is affected by the distributed constants of the line.

f. The equation $\beta=\frac{360° f}{V}$ also shows that the phase-shift constant, β, is proportional to the frequency, f, as described in *a*, *b*, and *c* above.

19. Group Velocity

In an ideal line, by definition, there is no attenuation, and the phase-shift constant, β, is directly proportional to the frequency of the transmitted wave. From the equation $\beta=\frac{360° f}{V}$, it can be seen that if β is proportional to f, then V must be a constant. (Looking at this conversely, if V is a constant, then β must be proportional to f.) Therefore, on an ideal line, the phase velocity, V, is the same for all frequencies. Now, consider the case of several frequencies on the same line.

a. Ideal Line. Assume that a complex wave composed of more than one frequency moves along a line on which the phase-shift constant is directly proportional to the frequency. Such a case is illustrated by two sine waves, f_1 and f_2, in A, figure 27. Adding these waves point by point yields the resultant wave, F. The two component waves, bearing the phase relationship shown as t_1, move down the line a distance d_p. At t_2, the same phase relationship still exists; that is, the component waves, of different frequency, move down the line at the same velocity. Thus, the resultant is unchanged, since the phase relationship between the two component waves is still the same at t_2. Thus, the envelope of the wave travels a distance d_g identical with d_p. By definition, the velocity with which an individual wave moves down the line is termed the *phase velocity*. The velocity with which the resultant, F, moves down the line is termed the *group velocity*. On the *ideal* line, where the phase-shift constant varies directly as the frequency of the transmitted wave, the resultant and the component waves will have the same velocity. In such a case, there is no separate group velocity; it is equal to the phase velocity.

b. Actual Line.

(1) Consider the case of an *actual* transmission line. Because of its distributed constants, the phase-shift constant is *not* directly proportional to the frequency. Therefore, waves of different frequency do not move along the line with the same velocity. Rather, the wave of highest frequency moves with the greatest velocity. This is because of the skin effect which reduces the inductance, and therefore the inductance reactance, at the higher frequencies. In B, figure 27, the transmitted waves, f_1 and f_2, and the

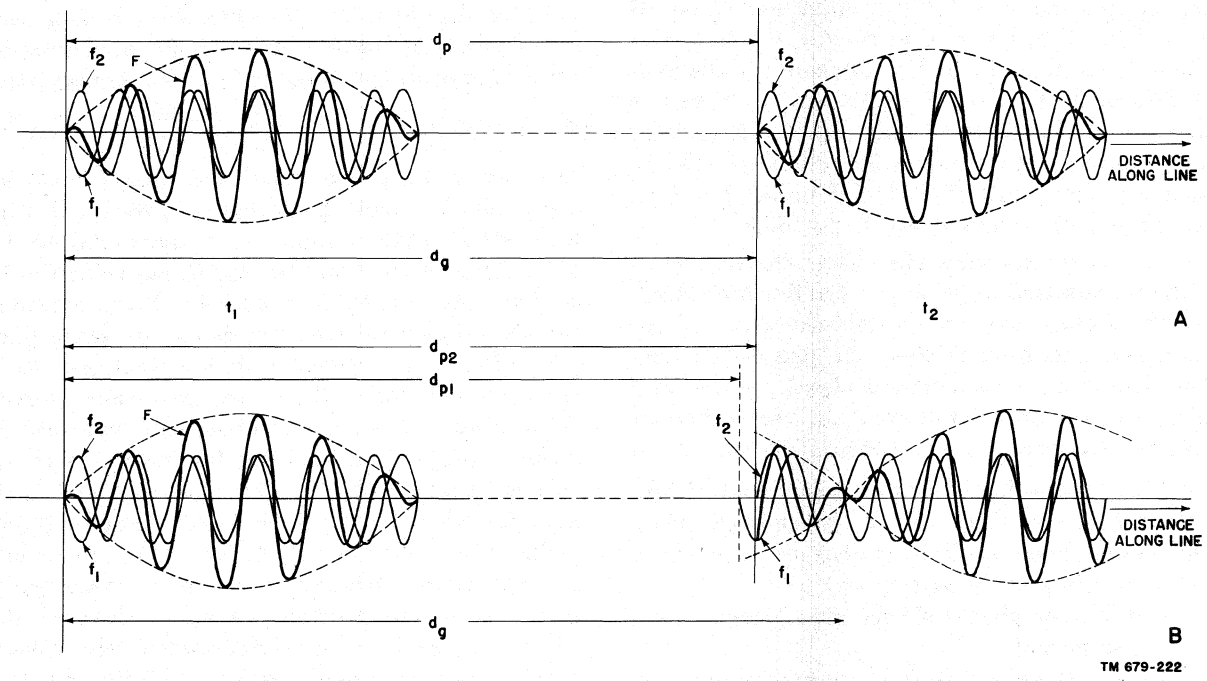


Figure 27. Phase velocity and group velocity.

resultant wave, F, start with the relationships shown in A. However, at a later time, t_2, wave f_2 has traveled a distance d_{p2}, but f_1 has traveled a shorter distance, d_{p1}. Thus, the phase of f_1 has shifted behind that of f_2. When f_1 and f_2 are added, the resultant, F, has its maximum amplitude at a point different from that on the ideal line. Its envelope, denoted by the broken line, has progressed a distance d_g, now greater than d_{p1}, or d_{p2}. It has thus moved farther and faster than either of the component waves, f_1 and f_2. The velocity of the envelope is the *group velocity* of the complex wave. When it is not the same as the phase velocity, the group velocity is always greater than the phase velocity of any component of the complex wave on the transmission line. (For wave guides, the reverse is true.)

Although the group velocity may be thought of as the velocity of the envelope of the wave, it is not the velocity with which the intelligence is conveyed by a complex signal, such as an amplitude-modulated carrier. The intelligence is conveyed by the sidebands, at a velocity which is the phase velocity of the sideband components. Neither the energy of the wave, nor the intelligence it represents, can travel faster than the fastest component of the wave.

(2) It must be pointed out that, for simplicity, the attenuation of the waves has been ignored. Attenuation does occur, however. As a matter of fact, the effects of attenuation and phase shift upon the transmitted signal are combined into a single constant, γ, called the *propagation constant* where $\gamma=\alpha+j\beta$ with α and β representing the attenuation and phase shift. The symbol j represents an angular rotation of 90° counterclockwise. The propagation constant, γ thus describes the effect that the transmission line has on the transmitted wave.

20. Distortion in Line

a. Desired Characteristics. As a wave travels along a transmission line, its amplitude is attenuated and its phase is shifted. For the wave to be transmitted without distortion, it is necessary that each component wave be attenuated in the same proportion, and that the phases of the component waves be shifted by amounts directly pro-

portional to their frequencies, no matter what the frequency components may be.

(1) Figure 28 illustrates the motion of an undistorted wave. The dotted line in A represents a 500-cycle wave with an amplitude of 10 volts. The dot-and-dash line represents a 1,500-cycle wave with an amplitude of 8 volts. The solid line represents the resultant input transmitted wave which is composed of the 500-cycle and 1,500- cycle waves.

(2) B, figure 28, represents the same wave after it has passed through a transmission line that has an attenuation ratio assumed to be 2 to 1. As a result, the 500-cycle wave (dotted line) now has an amplitude of 5 volts as compared with the 10 volts it had before transmission. Likewise, the 1,500-cycle wave (dot-and-dash line) is attenuated from 8 volts to 4 volts, which is a ratio of 2 to 1. Thus, the amplitude of the transmitted wave (solid line) also has been attenuated; but note that its shape remains the same; that is, the wave is undistorted.

(3) Note also that the 500-cycle wave, in B, has had its phase shifted by an amount assumed to be 30°. The 1,500-cycle wave must have its phase shifted by 90° in order that the phase shifts have the same proportions as the frequencies. This is what occurs in this case: 1,500:500=90°:30°=3:1. The phase of the resultant transmitted wave (solid line) necessarily has been shifted also, but note that the shape of the wave remains the same; it has not been distorted by its motion along the line.

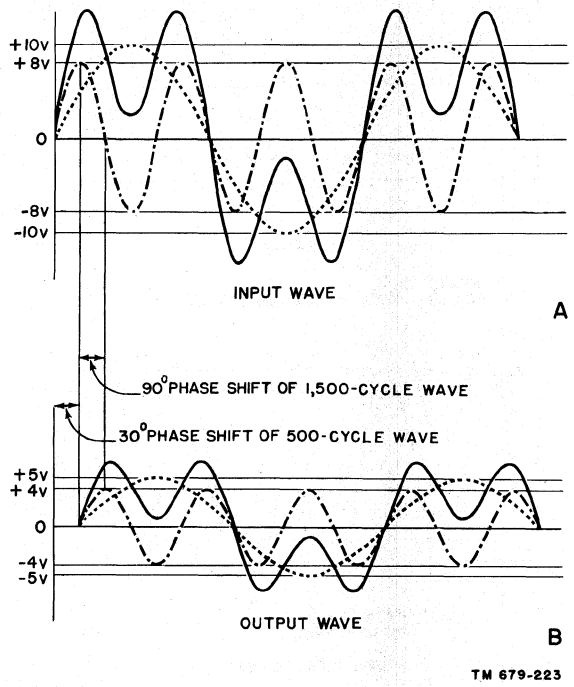


Figure 28. Desirable transmission conditions.

b. Frequency Distortion.

(1) On actual transmission lines, not all waves are attenuated the same amount; the attenuation *increases* as the frequency of the transmitted wave *increases*. The solid line curve in figure 29 demonstrates how the attenuation increases with frequency. This results from the effects of capacitance, leakage, inductance, and resistance of the line. The lower broken line represents the constant attenuation which is desirable.

(2) The effect of the variation in attenuation on the shape of the transmitted wave is represented in figure 30. The input wave, in A, is the same as that of A, figure 23. Again, after moving along the line some specific distance (B, fig. 30) the 500-cycle wave (dotted line) has an amplitude of 5 volts because it has been attenuated in the ratio 10:5, or 2:1. However, the 1,500-cycle wave (dot-and-dash line) now has an amplitude of only 2 volts. Because of the nonideal attenuation characteristics of the line, there has

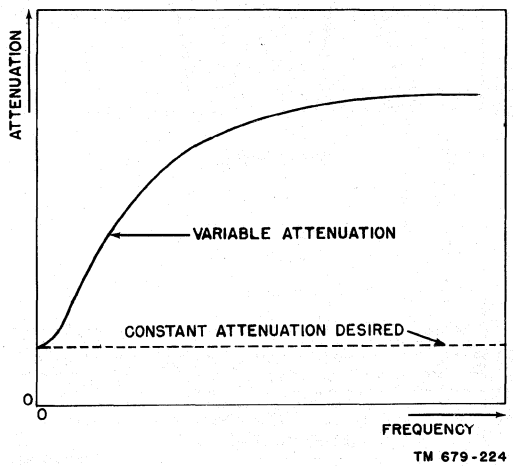


Figure 29. Variation of attenuation with frequency.

been a greater attenuation; in this example, a ratio of 8:2, or 4:1 has been assumed. Thus, the amplitude of the transmitted wave (solid line) also has been attenuated. Note, however, that the shape of the transmitted wave has been *changed:* that is, the wave has been *distorted.* This type of distortion, which is caused by the variation with frequency of the attenuation caused by the transmission line, is termed *frequency distortion* (or, sometimes, *amplitude versus frequency distortion*).

c. Phase Distortion. On actual transmission lines, the phase-shift constant does not vary directly as the frequency of the transmitted waves since the phase velocity varies with frequency. The broken-line graph in figure 31 shows how the phase-shift constant varies in the desirable, or ideal case—directly proportional to frequency. The solid-line graph shows how the phase-shift constant deviates from the ideal in the actual case —more at low frequencies than at higher frequencies. This results from a change in the values of reactance of either the distributed series inductance or the shunt capacitance. These distributed constants affect the phase velocity, V. And, since the phase-shift constant $\beta=\frac{360°f}{V}$, it is evident that the phase-shift constant, β, will also change as the frequency, and therefore, the reactance of the distributed change. The effect of this variation of the phase-shift constant is illustrated in figure 32. Again, the input wave (A, fig. 32) is identical with that of A, figure 28. In B, figure 32, the 500-cycle wave (dotted line) has had its phase shifted 45° lagging. For no distortion, the phase shift of the 1,500-cycle wave should have been 135° lagging. However, since the 1,500-cycle wave travels faster than the 500-cycle wave, the 1,500-cycle wave lags less than 135°, and in this case the 1,500-cycle wave (dot and dash line) has

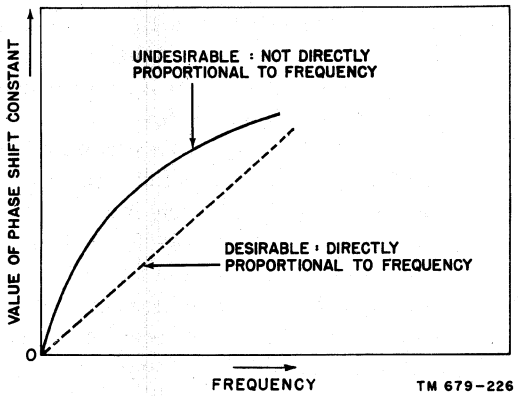


Figure 31. Variation of phase-shift constant with frequency.

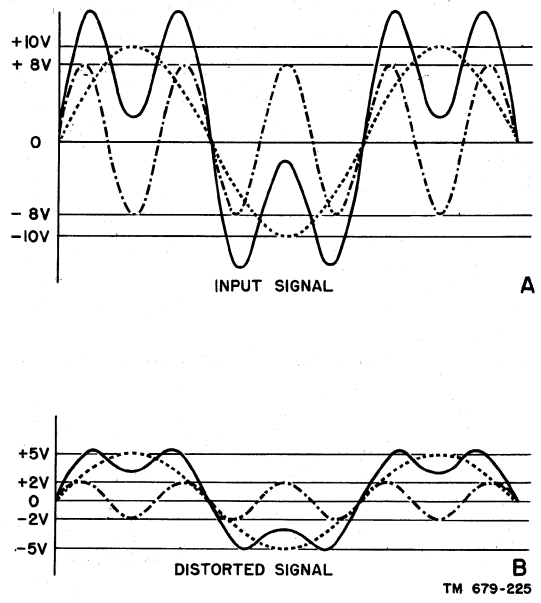


Figure 30. Frequency distortion.

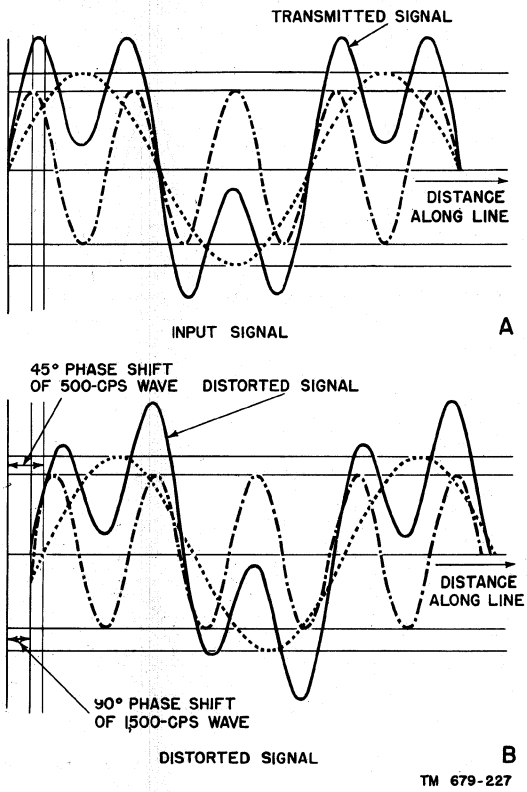


Figure 32. Phase distortion.

had its phase shifted only 90° lagging. The phase of each component of the transmitted wave has been shifted, but not by the desired amount. As a result, the shape of the transmitted wave (solid line) has been changed in its motion along the line; that is, the wave has been distorted. This type of distortion, which results from a phase-shift constant not directly proportional to the frequency of the transmitted wave, is called *phase distortion.*

21. Reflection in Line

Another effect of the transmission line on the transmitted wave results from *reflection.* Reflection is not caused by the properties of the line, however. It may result from a mismatch when equipment is inserted into the line, or it is caused by a sudden change in the properties of the line such as would result from inserting a cable into an open-wire line. Reflection is caused also by improper matching of the load impedance to the line.

a. Total Reflection. Maximum power is transferred to the receiver at the end of the line when the input impedance of the receiver is equal to the characteristic impedance of the line (par. 15). Consider what happens if a short circuit develops across the receiving end of the line, or if the load impedance *opens up.* Power is the product of voltage and current, EI. When the load impedance is shorted, the voltage across the short is equal to zero. When the load impedance is open, the current through the open circuit is zero. In either case, the power transfer to the receiver is zero. The power that travels along the line from the transmitter at the sending end must be reflected back to the transmitter, since it is not absorbed. When the power is transmitted to the receiver, it is in the form of current and voltage waves. When the power is reflected back to the transmitter, it must also be in the form of current and voltage waves. In other words, if reflection takes place at the receiver (or at any other point on the line), because of an impedance mismatch, there will be present on the line one pair of current and voltage waves traveling toward the receiver, and other pair traveling back toward the transmitter.

b. Partial Reflection. A similar condition exists when the input impedance of the receiver is not equal to the characteristic impedance of the line. In this case, only part of the transmitted power will be absorbed by the receiver; part will be reflected bact to the transmitter. The closer the value of the receiver impedance is to the characteristic impedance of the line, the greater will be the power absorbed by the receiver, and the smaller will be the power reflected back to the transmitter.

c. Need for Matching Impedances. In actual practice, the input impedance of the receiver is matched to the characteristic impedance of the line. The points at which reflection usually occur are at the junction of an open-wire line and a cable, at the junction of two different gages of wire, and at the input to equipment that has been inserted in the line without matching the input impedance of the equipment to the characteristic impedance of the line. In the first two cases, reflection results from the sudden change in the properties of the line. In the third case, reflection is caused by the mismatch between the equipment and the line. Reflection must be guarded against on long lines for two reasons:

(1) It reduces the power delivered to the receiver.
(2) The resulting reflected wave, or echo, lowers the quality of the transmitted signal by distorting the shape of the wave.

22. Summary of Transmission-Line Theory

a. When a signal is applied to a transmission line by a transmitter, the transmitted power travels along the line in the form of current and voltage waves.

b. The properties of a long transmission line are distributed uniformly over the entire length of the line, and are called the distributed constants of the line. There are four such properties—series resistance, series inductance, shunt capacitance, and shunt conductance. The effects of the transmission line on the transmitted wave are determined by the properties of the line.

(1) The presence of series resistance and shunt conductance in the line reduces the amplitudes of the current and voltage waves, causing attenuation of the power transmitted.
(2) When a wave moves a distance along a line, its phase is shifted a certain number of degrees, producing a phase difference between the input and output signals. The number of degrees the phase is shifted in a unit distance is called the phase-shift constant.

c. The velocity with which the wave moves along the line is called the phase velocity. This

is the velocity of a specific point or phase of the wave, and it is desirable that this velocity be constant. If it is not, waves of different frequencies will move down the line with different velocities. The velocity of the envelope is called the group velocity.

d. If the line causes signals of different frequencies to be attenuated by different amounts, the result is called frequency distortion. If the line causes signals of different frequencies to be shifted in phase by amounts not directly proportional to the frequencies, the result is called phase distortion.

e. Every transmission line has a characteristic impedance the value of which depends on the distributed properties of the line.

(1) This is the impedance across the input terminals of a theoretical line of infinite length. In practice, it is the impedance across the input terminals of any very long line.

(2) It is the input impedance of any length of line when the line is terminated by an impedance equal to the characteristic impedance of the line.

f. The purpose of a transmission line is to transfer maximum power from the transmitter at the sending end to the receiver at the receiving end. This may be achieved by making the impedances of the transmitter and the receiver equal to the characteristic impedance of the transmission line between them.

g. If, at any point on the line, the impedance is not equal to its characteristic impedance, some of the transmitted power will not be transferred to the receiver, but will be reflected back to the transmitter. This reflection of current and voltage waves, traveling back toward the transmitter, must be avoided for two reasons:

(1) Reflection reduces the power delivered to the receiver.
(2) The reflected wave distorts the transmitted signal.

Section II. TRANSMISSION LINE PRACTICE

23. Classification of Transmission Lines

Transmission lines usually are classified according to their construction. A distinction is made between open-wire lines and the different types of cables. It is important for the technician to know where these different types of transmission lines are used. He should understand also the construction and characteristics of different types of lines, and realize the correlation between line characteristics and individual applications. Three main types of transmission lines are described in paragraphs 24, 25, and 26—open-wire lines, cables, and field wires.

24. Open-Wire Lines

Open-wire lines have relatively low losses at voice and carrier frequencies. They operate satisfactorily over great distances; a round figure of 1,000 miles may be given.

a. Open-wire line is used for both carrier and voice frequency transmission. It generally consists of two open wires. Figure 33 illustrates how the line usually is carried on poles and suspended on a special high-dielectric glass insulator. The choice of spacing between the two conductors of a pair, and between separate pairs, depends on the carrier frequency. The higher the frequency, the smaller is the spacing between the conductors of a pair, and the greater is the spacing between pairs. The conductors usually are made of copper, or an alloy of copper and steel. The wire size of the conductors generally is 165, 128, or 104 gage, which means that the wires are 165-, 128-, or 104-thousandths of an inch in diameter.

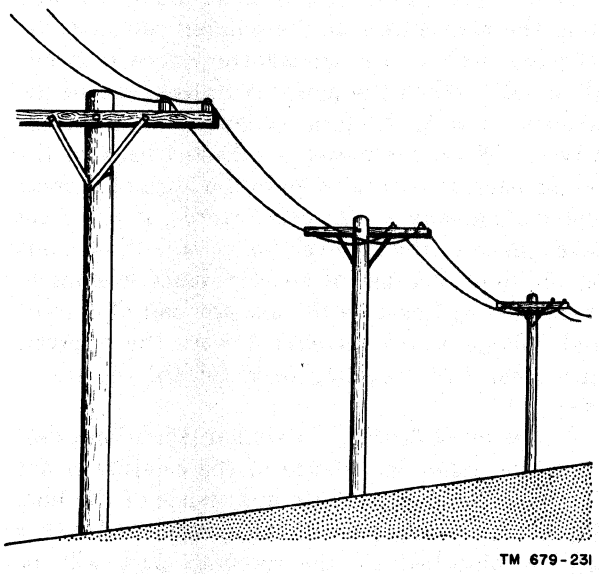


Figure 33. Open-wire line.

b. Signal attenuation produced by open-wire line is very low as compared with that produced by cables and field wires; it varies between .035 and .085 db (decibel) per mile, depending on the frequency, the wire size, the weather conditions, and the type of circuit. As the frequency increases from 500 to 5,000 cycles, the amount of attenuation increases by about 50 percent. However, the signal attenuation increases by about 200 percent when the frequency increases from 5,000 to 50,000 cycles. At a carrier frequency of 140 kc, the amount of attenuation is approximately 0.38 db per mile. Loading (a method for reducing attenuation by increasing the line inductance) (par. 29) never is used with open-wire lines, because adverse weather conditions affect open-wire loading detrimentally. The attenuation characteristics of the line change with weather, the amount of attenuation increasing in wet weather. Repeaters, which amplify the signal, are placed along the line when it is desired to increase the length of the telephone circuit. Because the attenuation on an open-wire line is low, the repeaters can be spaced relatively far apart, making the open-wire line a less expensive form of transmission system to install. Maintenance, however, is more expensive than it is for cable, especially under combat conditions. Table I lists different types of open-wire lines with their various characteristics.

Table I

Open-wire lines and their characteristics (based on standard spacing of 8 inches)

Wire size	D-c resistance (ohms per loop mile)	Capacitance (μf per loop mile)	Attenuation at 1 kc (db per mile)
#12 gage (104)	10. 15	0. 00905	0. 070
#10 gage (128)	6. 74	. 00944	. 049
#8 gage (165)	4. 11	. 00996	. 032

Notes.
1. All values are for dry weather conditions.
2. All measurements made at 68° Fahrenheit.
3. Number in column 1 is diameter of wire in thousandths of an inch.

25. Cables

Although it is customary to distinguish between lead-covered, rubber-covered, and submarine *cables*, the three types are basically the same in construction. Some cables, although not all, contain insulated conductors in the form of quads. The cable conductors are transposed by twisting the 2 wires of each pair together, by twisting the pairs of each group of 4 wires together to form quads, and by spiraling the quads in opposite directions about the core of the cable. The number of quads varies from seven (14 pairs) to 150 (300 pairs) per cable, or more. American manufactured cables use the multiple twin quad, or *bell* quad, which consists of two twisted pairs, twisted together in a long twist. Cables of foreign manufacture use either the multiple twin quad, or a spiral-four quad, also called a *star* quad. This type consists of four wires laid together and twisted as a group in a long twist, the diagonally opposite wires being used as a pair. The conductors are usually 19- or 16-gage solid copper. Some long circuits use each pair of a quad as a single conductor to decrease the signal attenuation. On short, local circuits, two-wire operation is permissible—each wire of a pair being used as a separate conductor. Table II lists various types of cables with their characteristics.

a. Lead-Covered Cable. This type of cable may be used in a permanent installation to provide a comparatively large number of voice-frequency circuits. It may be laid above ground or buried. Other applications of this type include use as an entrance cable between the end of an open-wire line and office equipment, or as an intermediate cable between two sections of an open-wire line. The cable consists of many quads, up to 150 or more, encased in a flexible lead pipe. The close spacing between the conductors of a pair causes an excessive attenuation. The attenuation varies between 2.0 to 3.8 db per mile at voice frequencies, depending on the size of the wire. The use of loading coils (par. 29) decreases the attenuation so that it varies between 0.15 and 0.5 db per mile. At carrier frequencies, it is not customary to use loading. Instead, closely spaced repeaters are used along the line, to make up for the relatively large amount of attenuation.

b. Rubber-Covered Cable.

(1) In this type of cable, the insulation around the individual wires is rubber, or a rubber substitute, as is the covering over the quads. Being comparatively light and flexible, rubber-covered cables may be installed aerially, laid on the ground, or buried. The use of rubber insulation prevents the absorption of moisture by the paper insulation, which would increase the attenuation. Cable Assembly CC–358, diagramatically illustrated in figure 34, is an example of rubber-

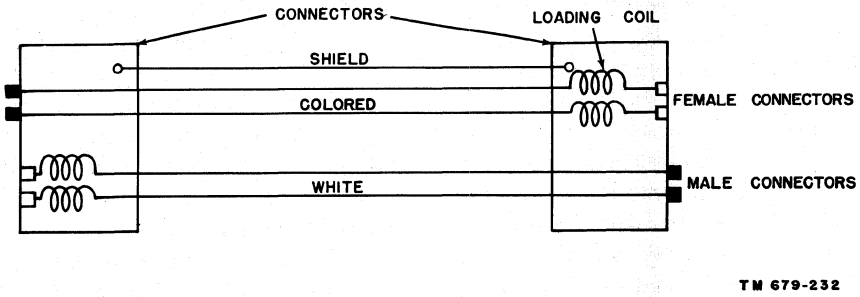


Figure 34. Cable Assembly CC–358.

covered cable. It consists of four rubber-insulated, stranded copper conductors twisted to form a spiral-four quad. The rubber insulation of one pair of conductors is white; the insulation of the other pair is colored. A basket-weave steel-wire braid covers the paper, and a rubber jacket covers the entire core. The cable is available in ¼-mile lengths with male and female connectors at opposite ends for rapid interconnection. Each connector assembly contains a 6-millihenry loading coil. This cable is used for four-channel carrier operation over intermediate distances; it also may be used aerially over distances up to 150 miles.

(2) Spiral-four Cable Assembly CX–1065/G is made from spiral-four Cable WM–8/G, which is a recent development. It is smaller and lighter than the previous spiral-four cable, and is also capable of use over wider frequency and temperature ranges. In the construction of WM–8/G cable, plastic materials have been used instead of rubber for the installation and the jacket (fig. 35). Four stranded-copper conductors are insulated separately with polyethylene and cabled around a ploye thylene core. The cabled conductors are covered by an inner jacket of polyethylene, a carbon-cloth stabilizing tape, a stainless steel braid, and a thermoplastic outer jacket.

(3) Cable Assembly CX–1065/G consists of approximately a ¼-mile length of Cable WF–8/G with a universal connector at each end. Loading coils are not built into the connectors as they were in earlier versions of spiral-four. Instead, Telephone Loading Coil Assembly CU–260/G may be inserted between the connector faces when required by the carrier system design.

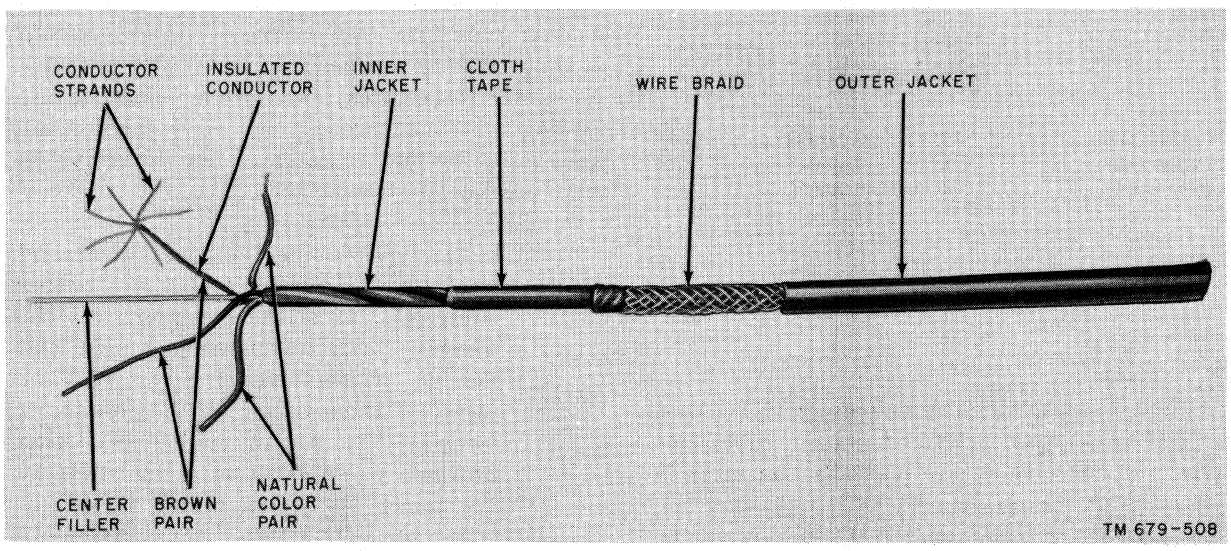


Figure 35. Cable WM–8/G, showing construction details.

(4) Rubber-covered cables also come in 5- and 10-pair types for use in voice frequencies. An example of 5-pair rubber cable is shown in figure 36. These are used principally for short loops where a number of circuits are required. Such cables are available with or without connectors at the ends. The conducting, or semiconducting, sheath about the cable *stabilizes* it. Cables are said to be nonstabilized if their distributed characteristics, capacitance, conductance, and resistance increase by large amounts from dry to wet weather. The sheath reduces these changes. Five- and ten-pair cables are fairly stable when in good condition, provided water does not get into the conductors.

c. Submarine Cable. The submarine cable is used as an insert in either open-wire lines or cable systems when the line must traverse a body of water too wide to be spanned aerially. The cable consists of paper-insulated conductors, lead-covered and wire-armored—that is, encased in a wire mesh. It is suitable for use in water to depths of 250 feet without danger of collapse of the sheath. Excessive water cannot penetrate along the cable even though the sheath should fail. In depths greater than 250 feet, there is danger of gradual collapse of the sheath; also, excessive water pressure may push the wires through the insulation, causing gradual failure of the cable. Submarine cables are made with extra-dense cores and thicker sheaths for use in depths up to 1,000 feet. Rubber-covered cables also may be used for submarine applications if there is no danger that the cable will be damaged by excessive abrasion. When rubber-covered cable is used under water, it is important that all connections and splices be made watertight.

Table II

Cable conductors and their characteristics

Size of cable conductor	Resistance (ohms per loop mile)	Capacitance (μf per loop mile)	Attenuation at 1 kc (db per mile)
#26 gage	440	0. 069	2. 67
#24 gage	274	. 072	2. 14
#22 gage	171	. 082	1. 79
#19 gage	86	. 084	1. 26

d. Coaxial Cables.

(1) This type of cable consists of an inner conductor and an outer conductor separated from one another by rubber or fiber disks (fig. 37). The characteristics of such cable—distributed constants and characteristic impedance—are determined by the sizes of the conductors used. The advantage of this type of cable is that the current flow is restricted to the outer surface of the inner cable and the inner surface of the outer cable. Interference external to the cable cannot penetrate to the interior of the cable where the current flow takes place. As a result, the outer cable acts as a shield against external electrical interference. The signal attenuation of coaxial cable is great, however—approximately 8.6 db per mile at 1 mc (megacycle). Therefore, repeaters must be spaced closely when the line is long.

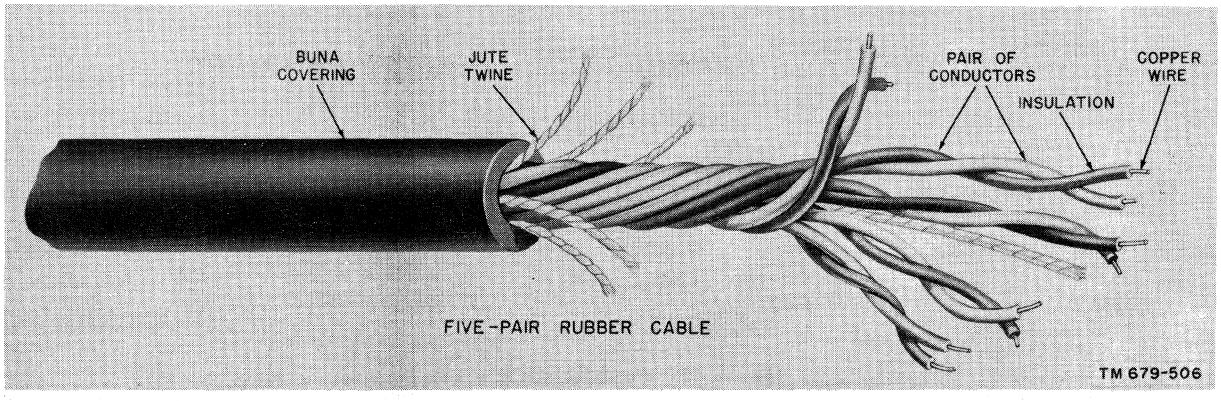


Figure 36. Five-pair rubber cable.

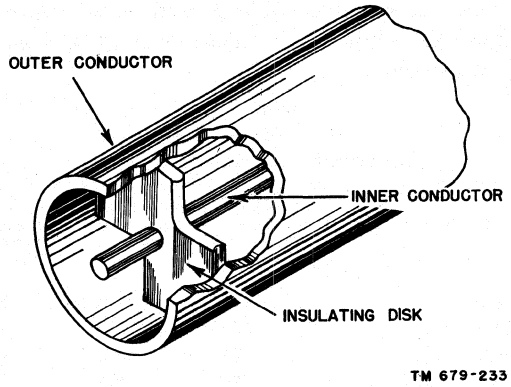


Figure 37. Cutaway view of coaxial cable.

(2) Figure 38 shows a typical cable assembly, containing 2 (there can be 4 or even 8) coaxial cables, in addition to several conventional pairs used for supervisory purposes. In carrier service, as many as 480 two-way telephone conversations may be carried simultaneously on any one such line.

26. Field Wires

Field wires are a special type of line designed for use in situations where communication lines must be set up at great speed. Desirable transmission properties are sacrificed in favor of light weight and strength. The line consists of a twisted pair made of strong, light conductors insulated with rubber or a rubber substitute. Signal attenuation on this line is high: 1.5 db per mile when the wire is dry, 2.5 db per mile when the wire is wet, at a frequency of 1,000 cycles. Field wire lines, in general, are operated both without loading and without repeaters, so that their use is restricted to short distances. The capacitance, leakage, and loss of this wire increase considerably with a change from dry to wet weather. The usable range may be increased, in some cases, by use of loading or voice-frequency repeaters. Repeater design for such lines is difficult, however, because of the sharp increase in attenuation when the wire becomes wet. Therefore, field wires are used for voice-frequency circuits where an appreciably long talking range is not required. Figure 39

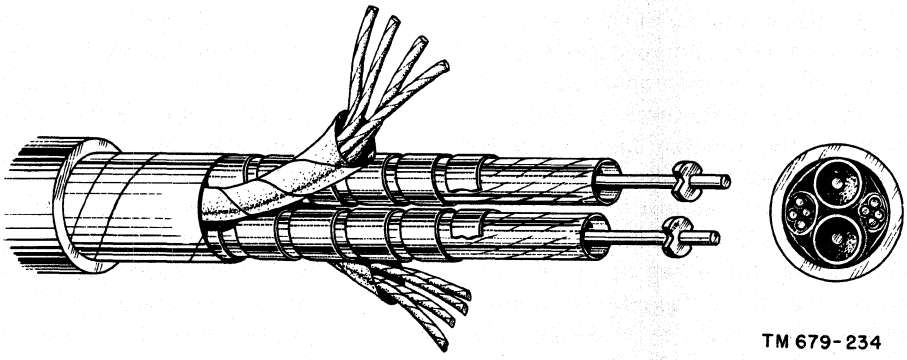


Figure 38. Typical coaxial-cable assembly.

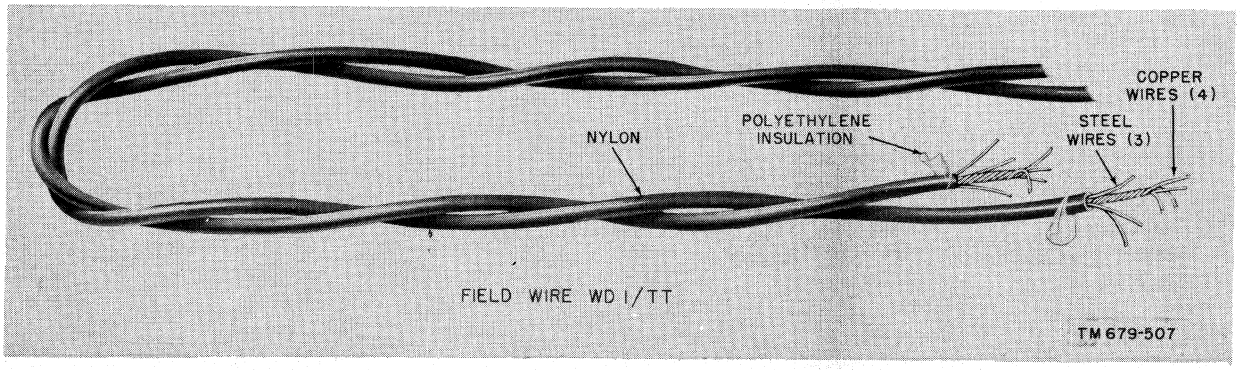


Figure 39. Field Wire WD 1/TT.

depicts field Wire WD 1/TT, and gives an idea of the nature of its construction.

27. Line Problems

a. Attenuation in Line. The primary purpose of a transmission line is to transfer maximum power to the receiver at the receiving end of the line. Achievement of this purpose involves two separate factors. First, the line itself must absorb as little power as possible, delivering maximum power to the receiving end. Second, the power that arrives at the receiving end should all be absorbed by the load, and there should be no reflection back along the line. The latter requirement is met by proper matching of the load to the line. The former is met by keeping *attenuation* of the signal by the line at the lowest possible value.

b. Interference in Line. In communication circuits, it is also necessary that the transmission line transmit *only* the desired signal. Undesired electrical signals—interfering signals—can be picked up by the line in a variety of ways, and transmitted along with the desired signal. These interfering signals may distort the desired signal, or even override it so that it is rendered unintelligible at the receiver. Methods of avoiding such interference, or reducing it to minimum, are discussed in paragraph 36*g*, *h*, and *i*.

28. Causes of Attenuation

The characteristics of a transmission line which attenuate the signal may be summarized as follows:

a. A match, or mismatch, between the impedances of transmitter and receiver and the characteristic impedance of the transmission line, has an influence on how much power is transferred from the transmitter to the receiver.

b. Reflection of power because of a mismatch at the joint of two dissimilar line sections will result in a decrease of power transferred to the receiver.

c. Power losses will occur in pieces of equipment that are part of the communication circuit. Such losses are not really line losses, but they may be reduced by proper choice and maintenance of the line. The power losses due to mismatches between line and equipment may be reduced to a minimum by careful matching; those in the equipment itself cannot be avoided.

d. The attenuation per unit length of line, and the length of the line, determine the total line attenuation. To this must be added the losses inherent in equipment, in order to obtain the total attenuation of the circuit.

e. If the characteristic impedance of the line varies with frequency, the match between line and receiver is not constant and there is a variation in the power transfer.

f. Frequency distortion reduces the power transfer by amounts varying with frequency, reducing the power of the higher frequencies more than the lower frequencies. Consequently, a signal containing components of different frequencies will have its waveshape distorted. Obviously, these power losses may be reduced only by altering the characteristics of the transmission line.

29. Loading of Line

a. Function of Loading. The *loading* of a transmission line changes its characteristics and thus reduces the total attenuation, causing the characteristic impedance and attenuation per unit length to remain substantially constant over a desired frequency range.

b. Effect of Loading.

(1) Which of the four distributed constants of the transmission line can be changed for loading purposes? Reducing the distributed resistance lowers the attenuation, but it may be accomplished only by increasing the size of the wires. This in turn increases the amount of copper used and thus decreases the number of circuits per cable. Reducing the distributed conductance has little effect on the attenuation. Reducing the distributed capacitance reduces the attenuation, but it may be accomplished by increasing the separation between the conductors. This also reduces the number of circuits per cable. The solution of the problem lies with the fourth constant: the series inductance. In practice, the value of this inductance is increased deliberately, whereas the other three constants are kept as small as possible. This corrective measure is called *loading*.

(2) The effect of loading on attenuation is illustrated in the following example: In A, figure 40, assume that 5 watts is the input power, P, to a transmission line having a characteristic impedance of 500 ohms. The ratio of input voltage and input current must be equal to the char-

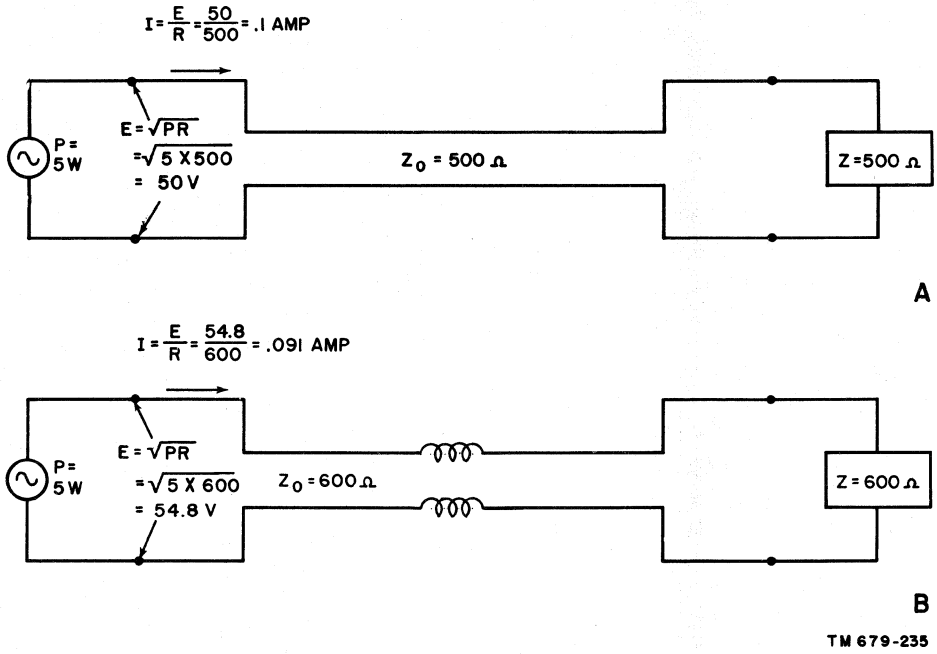


Figure 40. Effect of loading.

acteristic impedance, because the line is terminated with an impedance equal to the characteristic impedance of the line. As shown, the input voltage must be 50 volts, and the input current 0.1 ampere. Now, assume that the characteristic impedance of the line is increased by the inductive loading, as indicated in B, figure 40. The characteristic impedance is now 600 ohms. Since the input power is held constant, the input voltage is now 54.8 volts, and the input current 0.091 ampere. This reduction of current decreases the I^2R losses along the line, so that if loaded and nonloaded circuits receive the same amount of input power, more power is delivered to the receiver by the loaded circuit than by the nonloaded circuit.

(3) Figure 41 shows that the attenuation over a wide freqeuncy range is much less with loading than without loading. The same curves show that loading has caused the attenuation to be practically constant over a wide range of frequencies, whereas, without loading, the attenuation increases as the frequency increases. With loading, therefore, frequency distortion is virtually eliminated. Mathematically, it may be shown that loading causes *both*

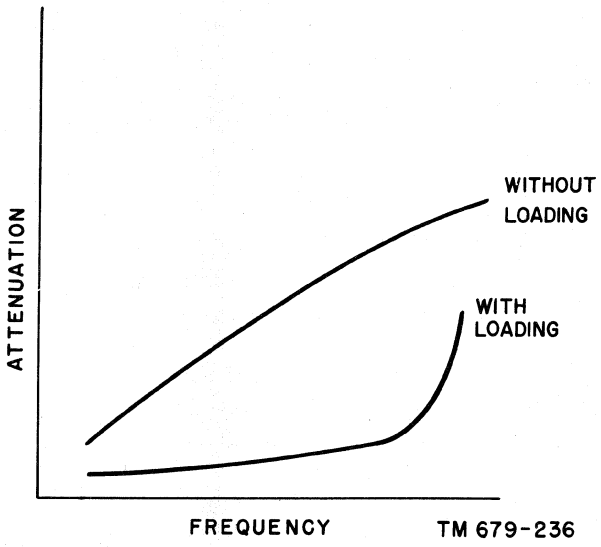


Figure 41. Effects of loading on attenuation.

attenuation and characteristic impedance to be substantially constant. With the characteristic impedance of the line constant, there is no variation in the match between receiver and transmission line. Hence, the power transferred to the receiver is constant, and maximum over a wider range of freqeuncies.

c. Methods of Loading. Loading may be accomplished in either of two ways. Both methods must provide an evenly distributed inductance

to give the effect of a transmission line with distributed inductance. With uniform loading, the complete length of line is wrapped with a tape of ferrous material, such as iron or permalloy. This method is expensive. The other method uses *loading coils* spaced at equal intervals along the line.

d. Loading Coils. Loading coils are so spaced that the change produced in a signal over a given distance differs little from the change that is produced over the same distance by a uniformly loaded circuit. The spacing must be such that there will be several coils per wavelength of signal; otherwise the attenuation will increase rather than decrease. The coils are wound on toroidal cores, which are made of powdered soft iron or powdered permalloy, with a binder to hold the powdered metal together. The air pockets in the powdered metal reduce the possibility of saturation of the core, which would distort the signal.

e. Limitation of Loading.

(1) The number of loading coils cannot be increased indefinitely because this would cause an increase in the series resistance which would cancel the beneficial effects of loading.

(2) Distortion cannot be completely eliminated in a loaded circuit. The effective resistance of a loading coil varies with frequency, as a result of hysteresis and eddy current losses in the core of the loading coil.

(3) The combination of distributed capacitance and lumped inductance acts like a filter. The curve for the attenuation of a loaded line in figure 41 shows that this equivalent filter causes the attenuation to increase sharply when the frequency exceeds a certain value, referred to as the *cutoff frequency*. Therefore, loading coils usually are designed to cover the band of frequencies *to be transmitted*, with the cutoff frequency occurring beyond the upper end of this band.

30. Interference in Lines

The theoretical discussion of transmission line characteristics up to this point has assumed that the lines under consideration are isolated—that is, that they are far from ground on other transmission lines or any type of current-carrying circuit. Actually, isolated transmission lines are rare. Two types of *interference* may occur between adjacent lines—*noise* and *crosstalk*. Both affect practical transmission line operation to an important extent.

a. Most lines are near other transmission lines, or are near the surface of the earth, which itself is a conductor referred to as *ground*. Communication lines operate at low power levels; that is, the lines are called upon to transmit relatively small amounts of power. As a result, the lines are susceptible to *noise* interference from other electrical circuits, particularly powerlines paralleling the communication lines. Most powerlines carry current at 60 cycles. The most troublesome interference arises from those harmonics of this frequency that fall in the 250- to 2,700-cycle audio range.

b. Interference is also produced by coupling between adjacent communication lines. Such interference is called *crosstalk*. The presence of crosstalk on a transmission line must be prevented, if possible, or at least reduced to a minimum, because of its effect on the intelligibility of a message. Crosstalk exists when electrical energy is transferred from one communication circuit to another, cuasing the conversation on one circuit to be faintly audible on the other.

31. Sources of Interference in Lines

a. Electrical Causes. Five common electrical conditions are responsible for interference.

(1) An unbalanced condition between the transmission lines and ground is the cause of noise induced by a powerline system.

(2) An unbalanced condition between pairs of transmission lines results in crosstalk resulting from electrostatic coupling between the lines.

(3) Electromagnetic coupling between the lines may produce crosstalk.

(4) Conductive coupling (which seldom occurs in well maintained lines) may cause crosstalk.

(5) A resistive unbalance may be still another source of crosstalk.

b. Location of Sources of Interference. Several of the electrical causes of interference can be present at more than one point in the transmission system. For places most likely to be sources of interference from one or more electrical causes, together with appropriate methods of preventing or reducing interference, see paragraph 33.

32. Electrical Causes of Interference in Lines

a. Unbalance to Ground.

(1) Distributed capacitance exists between the two conductors of a transmission line (par. 9). Similarly, there is distributed capacitance between ground and each conductor of a transmission line. This is shown in figure 42, in which C_1 and C_2 represent the capacitances between the conductors and ground.

(2) A transmission line is said to be *balanced to ground* if C_1 and C_2 are equal; that is, if the distributed capacitances to ground of the two conductors of the line are equal. This is the case when the two conductors are at equal distance from ground. Voltages are induced in the two conductors, through the distributed capacitances, by current flow in the ground. If the line is balanced to ground, the voltages induced in the two conductors are equal and opposite, and the potential difference between the conductors caused by the induced voltage is zero.

(3) If the line is unbalanced to ground, the induced voltages are unequal and there is a potential difference between the lines. This potential difference adds to the potential difference between the lines which results from the transmitted signal. It constitutes a new and unwanted signal added to the first, the waveform of which is thereby distorted. In other words, the unbalanced condition between the transmission line and ground is a source of interference. Since power-line systems cause a 60-cycle current flow in the ground, an unbalanced condition between a communication line and ground is the source of 60-cycle noise in the communication line.

b. Unbalance between Pairs. Interference can take place between adjacent transmission lines because of a capacitive unbalance between the lines themselves.

(1) A, figure 43, which represents 2 pairs of transmission lines, comprising 2 circuits, illustrates this condition. C_{13} is the distributed capacitance between conductor 1 of circuit I and conductor 3 of circuit II. Similarly, C_{14}, C_{23}, and C_{24} represent the other distributed capacitances between conductors of the

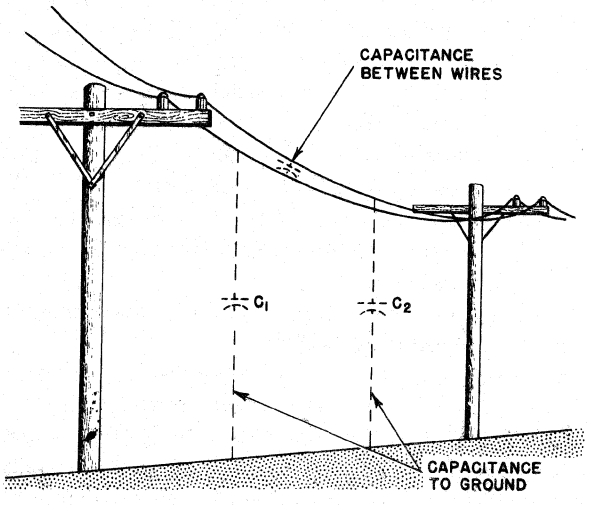


Figure 42. Distributed capacitance between transmission line and ground.

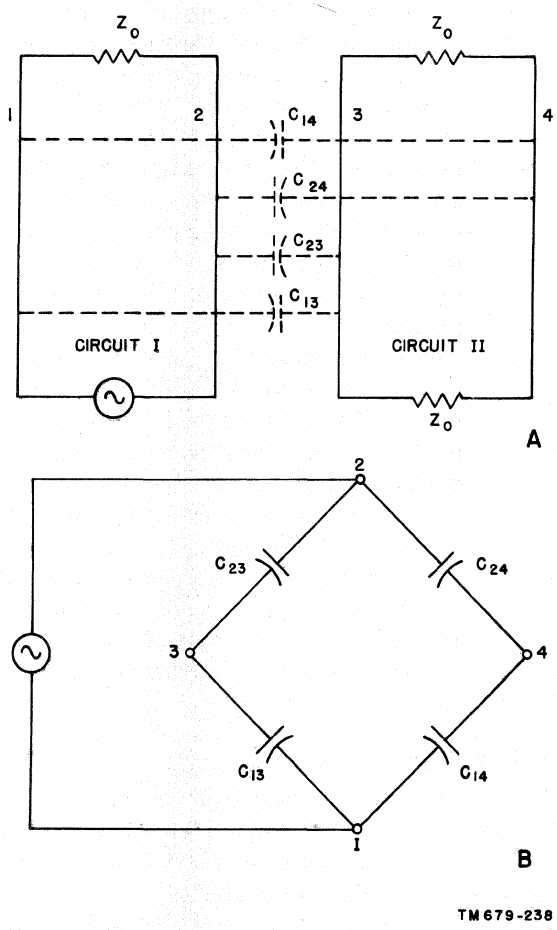


Figure 43. Distributed capacitances between adjacent conductors.

2 circuits. Energy may be transferred from one circuit to the other when the electric field of one circuit causes a voltage to be developed in the second circuit. For this reason, this type of unbalance is called *electrostatic coupling*, and it results in crosstalk.

(2) The equivalent circuit of the 2 circuits, or pairs of conductors, so coupled, is a bridge circuit (B, fig. 43). If $C_{13}=C_{14}$ and $C_{23}=C_{24}$, the voltage of lines 1 and 2 divides in the same ratio across C_{13} and C_{23}, and across C_{14} and C_{24}. Thus, equal voltages appear across equal capacitances, and no potential difference exists between 3 and 4 because of the voltage of 1 and 2. The circuit then is said to be balanced. Note that for balance to exist in the opposite direction—that is, no voltage coupled from 3 and 4 into 1 and 2—it is also necessary that $C_{13}=C_{23}$ and $C_{14}=C_{24}$.

(3) This balance is achieved by proper physical positioning of the lines, so that the capacitances have the equality desired. For example, an unbalance exists if the separation between conductors 2 and 3 is larger than the separation between conductors 1 and 4; under this condition, C_{23} is smaller than C_{14}, and a signal transmitted over circuit II will have added to it part of any signal simultaneously being transmitted over circuit I as a result of the variation of the electric fields between the conductors.

c. Electromagnetic Coupling. *Electromagnetic coupling* exists between 2 pairs when the magnetic field of 1 pair induces a voltage in an adjacent pair. Figure 44 represents two pairs of transmission-line wires. The varying current in the pair of circuit I sets up a magnetic field about conductors 1 and 2. This magnetic field spreads out and cuts conductors 3 and 4 of the pair of circuit II. If conductors 3 and 4 were at equal distances from conductor 2, they would be cut by an equal number of magnetic lines of force; the voltage induced in wires 3 and 4 would be equal, so that there would be no potential difference between wires 3 and 4, because of the current in conductor 2. However, conductor 3 is closer than conductor 4 to conductor 2. Thus, both conductors in the pair of circuit II will be cut by different numbers of lines

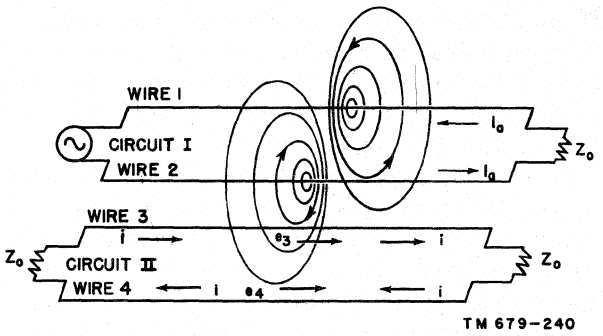


Figure 44. Effect of magnetic induction on crosstalk.

of force, so that different voltages, e_3 and e_4, will be induced in the two conductors. There is, then, a potential difference between conductors 3 and 4 as a result of the current flow in the pair of circuit I. This potential difference will be in addition to the potential difference between the conductors caused by the transmitted signal. The combination of the two potential differences results in an altered signal and a garbled message at the receiver.

d. Conductive Coupling. *Conductive coupling* exists when there is leakage between two pairs of conductors as a result of faulty insulation. Such coupling usually does not occur on well kept systems.

e. Resistive Unbalance. A *resistive unbalance* exists when the resistances in the two wires of a transmission line are unequal; it is caused by the presence of different gages of wire, improper splices, or poor connections. Referring to figure 44, it may be seen that if conductors 3 and 4 of pair II are at equal distances from conductor 2 in pair I, the induced voltages in conductors 3 and 4 are equal, and no crosstalk results. If the resistances in conductors 3 and 4 are unequal, however, the induced currents in the conductors are unequal. The difference in current is added to the current of the transmitted signal, resulting in crosstalk. The obvious remedy for a resistive unbalance is to balance the circuit.

33. Reduction of Interference in Lines

Interference reduces the quality of the transmitted signal by changing the shape of the signal. In a communication circuit, where the signal must be transmitted with minimum distortion, so that it does not lose its intelligibility, it is obviously necessary that noise and crosstalk on the transmission line be held to absolute minimum.

a. Noise from Powerline. *Noise* usually is produced by coupling between a circuit carrying a

strong current, such as a power line, and a circuit carrying a weak current, such as a communication line. The transmission lines of a communication circuit, therefore, must be well separated from the transmission lines of a power circuit.

b. Crosstalk at Receiving End. The sending end of one line must be kept well separated from the receiving end of another line. A transmitted signal is relatively strong at the sending end of a transmission line. Because of attenuation along the line, a signal is weak at the receiving end, and, unless the receiving end is well separated from the sending ends of other lines, there is likelihood of *crosstalk* at the receiving end.

c. Spacing of Pairs. Crosstalk between two pairs of conductors may be minimized by using a line configuration such that the effect of the magnetic field of one pair influences both conductors of the other pair equally. In figure 45, the conductors of the two pairs labeled GOOD are so placed that conductors 3 and 4 of one pair are at equal distances from conductors 1 and 2 of the second pair. The POOR configuration may result in crosstalk.

d. Spacing of Open Wires. Proper spacing of the two conductors of an open-wire line is effective in minimizing crosstalk. The separation between the two wires of a pair is kept small, relative to the separation between adjacent pairs. This is illustrated in figure 46, which shows that the wires of a pair are 8 inches apart, while adjacent pairs are 16 inches apart. The close spacing between wires of a pair causes both wires to be cut by approximately the same number of magnetic lines of force emanating from an adjacent pair. The smaller the difference in the number of lines of force cutting the individual conductors, the smaller is the difference in the voltages induced in the two wires, causing the crosstalk current to be small. However, the separation between the wires of a pair cannot be too small, since there is always the danger that wind might cause the two wires to touch each other, producing a short circuit. Spacing adjacent pairs relatively far apart reduces the number of magnetic lines caused by one pair cutting the adjacent pair. The induced voltages are then small, and so, consequently, is the crosstalk.

e. Crosstalk in Lines with Repeaters. Lines which use repeaters are extremely susceptible to interference. Interference tends to increase with the length of the line, the difference in signal strength in adjacent circuits, and the frequency of the signal. Repeaters are used on long lines to amplify the strength of the transmitted currents. Thus, lines using repeaters are more susceptible to noise and crosstalk. The crosstalk at the sending end can be eliminated almost completely by using separated circuits for the two directions of transmission, as shown in figure 47. The outgoing wires, A′ and B′, of circuits A and B, form a pair, as do the returning wires, A″ and B″. If crosstalk is induced from circuit A into circuit B at point Q (any point), it is not able to travel toward the sending end of circuit B because of the one-way action of the repeater. Lines A′ and B′ are at a high power level since they are at the sending end. Lines A″ and B″ are at a low power level since they are at the receiving end. The physical separation between the conductors of the outgoing and incoming circuits minimizes the possibility of crosstalk induced by this condition.

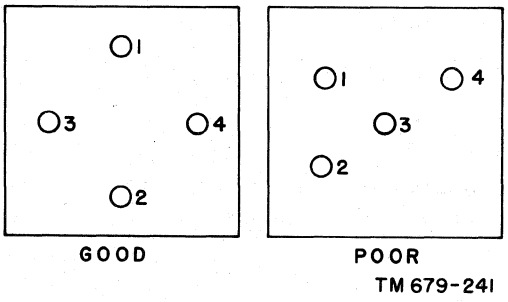


Figure 45. Configuration of wires to minimize crosstalk.

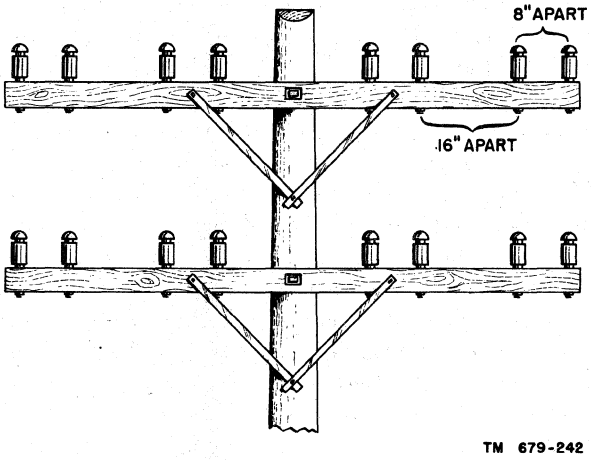


Figure 46. Spacing of wires to minimize crosstalk.

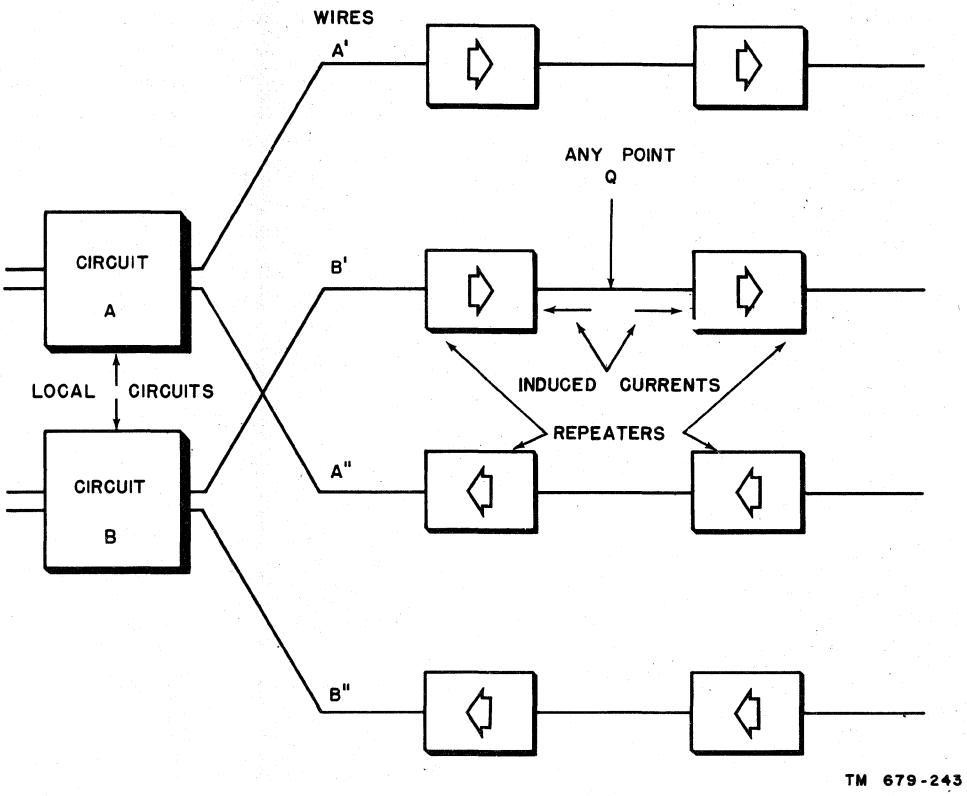


Figure 47. Spacing of conductors with repeaters to reduce crosstalk.

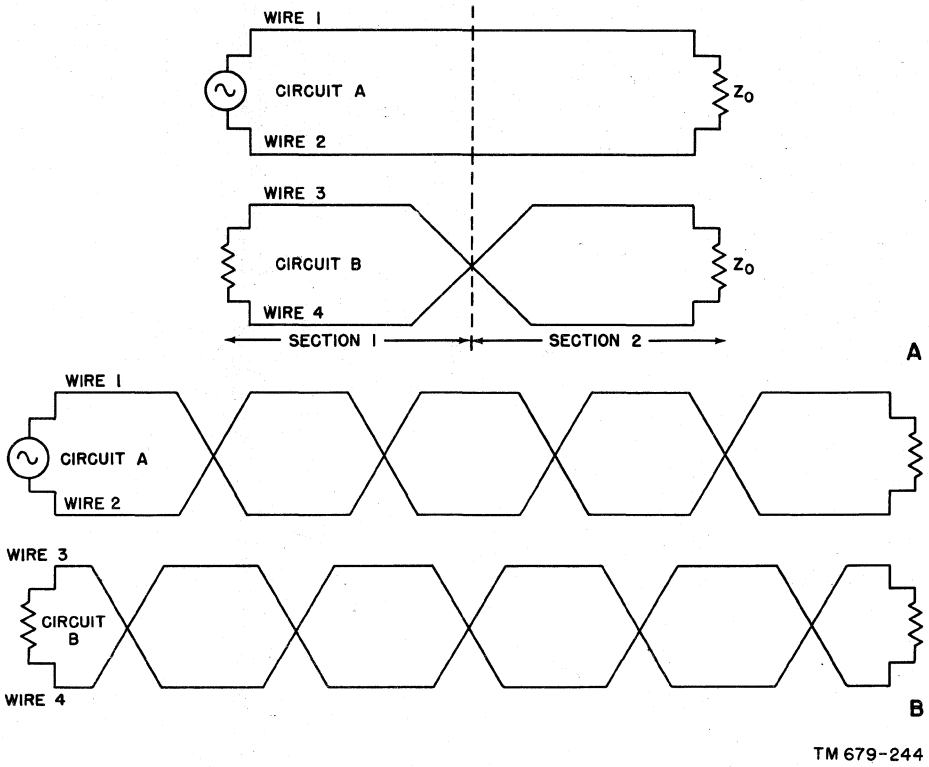


Figure 48. Transposition of wires to reduce interference.

f. Transposition of Wires.

(1) *Transposition* is a method of minimizing interference and is used on open-wire lines. The positions of the two conductors of each pair are transposed (interchanged) at specific intervals along the line. An example of a transposition is shown in A, figure 48. In the first line section, the varying current in the pair of circuit A induces a greater voltage in wire 3 of the pair of circuit B than it does in wire 4, since wire 3 is closer than wire 4 to the pair of circuit A. But, in the second section, conductors 3 and 4 have been transposed, so that the voltage induced in wire 4 is now greater than the voltage induced in wire 3. If the voltages induced in wire 3 in the two sections are equal and opposed to the voltages induced in wire 4, they cancel out completely and there is no crosstalk.

(2) Actually, the current in a line is not constant over its entire length, but decreases in amplitude because of attenuation along the line. As a result, the voltage induced in the first section does not equal the voltage induced in the second section. To compensate for this, the wires are transposed at frequent intervals along the entire length of the line. This causes the voltage induced in one transposed section to be more nearly canceled by the voltage induced in the adjoining section. Successive transpositions are illustrated in B, figure 48. The possibility of interference also may be reduced by transposing pair A, rather than pair B. The wires of both pairs must not be transposed at the same point, however, or the purpose of transpositions is defeated. Transposition intervals range from approximately 100 feet to 6 or 8 miles, depending on the frequency of the transmitted signal.

(3) Two common methods of accomplishing the transpositions are the point-type and drop-bracket methods. The point-type method, shown in A, figure 49, is used on lines for carrier systems because it does not change the configuration of the wires in adjacent transposition sections, as does the drop-bracket method, shown in B.

(4) The effectiveness of the transposition of conductors depends on the absence of reflected waves on the line. Reflected waves cause variations in the current amplitude along the line, and thus of the induced voltage. For this reason, poor splices or faulty insulation, which cause reflection, may produce crosstalk.

(5) Field wires, being wound spirally, are effectively transposed, and are relatively free of interference.

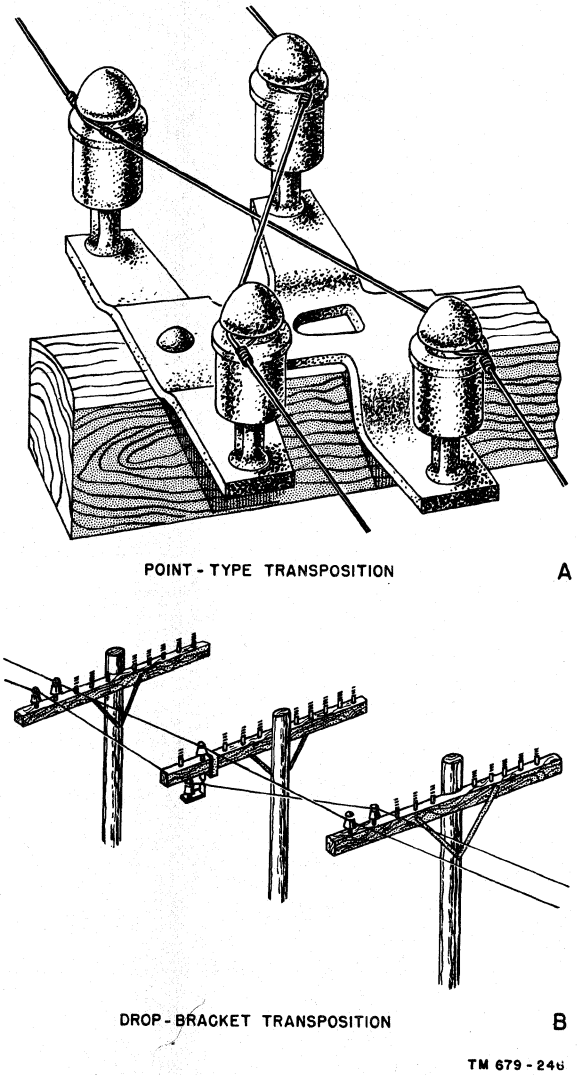


Figure 49. Methods of transposition on open wire line.

34. Minimizing of Crosstalk in Cable Circuits

a. The amount of crosstalk induced in cable circuits by electromagnetic coupling is negligibly small for two reasons—

(1) The wires in a cable are close to one another. As a result, there is little difference in the number of magnetic lines of force that cut across the two wires of a pair. Crosstalk caused by electromagnetic coupling is stronger on open-wire lines than on cables because of the greater separation of the wires of each pair in the former.

(2) The wires in a cable are transposed. The two wires of a pair are transposed by twisting them together. Two pairs of wires are transposed by twisting them to form a quad. As a result of the close spacing of the wires in a cable, and of the transpositions achieved by twisting the wires, crosstalk caused by magnetic flux is negligibly small at voice frequencies.

b. An important source of crosstalk in cables is electrostatic coupling between pairs of conductors (par. 32*b*).

(1) One satisfactory remedy for correcting the capacitance unbalance is to connect small capacitors across the lines (fig. 50).

(2) Another remedy is to divide the cable into sections. In splicing one section to another, a length of cable having a low value of distributed capacitance is spliced to a section of cable having a high value of distributed capacitance. Thus, the capacitance between the wires is made nearly equal, achieving better-than-average balance between the lines.

c. At carrier frequencies, both electromagnetic coupling and electrostatic coupling are important in cables. Separately shielded cables are used as the conductors for the two directions of transmission. The use of shields minimizes crosstalk caused by placing transmission lines at different power levels near one another. Noise, caused by 60-cycle powerlines, is of no consequence, because the shield of the cable effectively shuts it out.

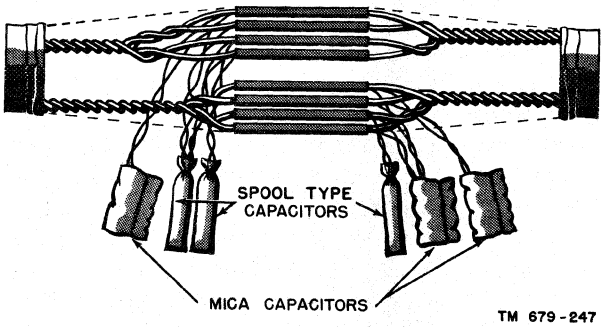


Figure 50. Installation of balancing capacitors at cable splice.

35. Phantom Circuit

In a communication network, the conductors of two circuits may be utilized in such a manner as to provide a third circuit with no increase in the number of conductors used. The 2 wires of 1 circuit act as a single conductor in 1 direction for the third circuit, and the 2 wires of the second circuit act as a single return conductor for the third circuit.

a. Consider the two communication circuits shown in figure 51. Conductors 1 and 2 form one circuit; conductors 3 and 4 form the second circuit. These are called the *principle circuits*, or *side circuits*. Using these same conductors, it is possible to include a third circuit. Wires 1 and 2 act as one conductor for the third circuit, and wires 3 and 4 act as the return conductor. Thus, there are 3 circuits using only 4 conductors. The third circuit is called a *phantom circuit*, and the combination of the 2 side circuits and 1 phantom circuit is called a *phantom group*.

b. Ordinarily, with one set of conductors as the transmission line for two different signals, there is danger of mutual interference. This possibility is eliminated by the careful balancing of conductors 1 and 2 and of conductors 3 and 4 (fig. 51). The balancing is achieved by connecting the conductors of the phantom circuit at precise midpoints on the side circuits. Then, current in the phantom circuit divides equally between conductors 1 and 2. The potential difference between points A and B is equal to the potential difference between points A and C. Similarly, the potential differences between points D and E and points D and F are equal. This means that between points B and C, and between points E and F, there is zero potential difference because of the current in the phantom circuit. There are potential differences between the points mentioned as a result of the side-circuit signal. Since, under ideal conditions, the current in the phantom circuit develops no potential differences in the side circuits, there is no interference between the separate signals.

c. A *resistive* unbalance (par. 32*e*) may cause crosstalk even though the conductors of the

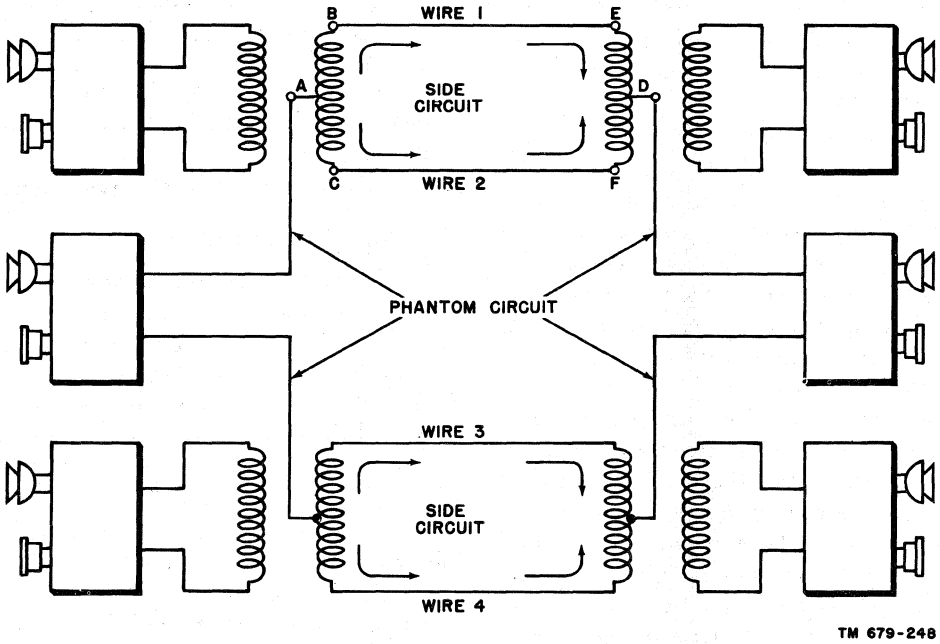


Figure 51. Phantom group.

phantom circuit are connected at the midpoints of the side circuits (fig. 51). The unbalance will cause the currents in wires 1 and 2 to be unequal as well as the currents in wires 3 and 4. Since crosstalk results when there are such potential differences in the circuit, resistive unbalance cannot be permitted.

36. Summary of Transmission-Line Practice

a. The open-wire line is carried on poles, suspended on glass insulators. The losses on this type of line are low at voice and carrier frequencies, and the line may be used over practically any distance.

b. A cable consists of insulated conductors, twisted together in pairs or quads. There may be as many as 7 to 150 quads (14 to 300 pairs) for each cable.

(1) The lead-covered cable is encased in flexible lead pipe. It is used above ground or buried, as an entrance cable between the end of an open wire line and office equipment, or as an intermediate cable between two sections of an open-wire line. Lead-covered cables may be used in permanent installations.

(2) In rubber-covered cables, the insulation around the individual wires and the covering over the quads is rubber or a rubber substitute, which makes these cables lighter and more flexible than lead-covered cables. The rubbered-covered cable may be installed aerially, laid on the ground, or buried.

(3) The submarine cable consists of paper-insulated conductors, lead-covered, and encased in wire armor. The cable is used as an insert in either an open-wire line or a cable, when the insert has to pass through a body of water too wide to be spanned aerially.

(4) A coaxial cable consists of an inner conductor completely surrounded by an outer conductor, with the two conductors insulated from one another by rubber or fiber disks. This cable has a high immunity to interference.

c. Field wires are a twisted pair of strong, light conductors, insulated with rubber or a rubber substitute. This type of cable is designed to be strong and of light weight, to be used in situations where there is a need for setting up communication lines at great speed.

d. The amount of power transmitted to the receiver is attenuated in the following ways:

(1) The series resistance and shunt conductance dissipate some of the transmitted power.

(2) The characteristic impedance of the line varies with frequency, causing the power transferred to the receiver to vary.

(3) The attenuation caused by the line varies with frequency, so that varying amounts of power reach the receiving end of the line.

(4) The impedances of the receiver and transmitter are not matched to the characteristic impedance of the transmission line.

(5) Reflection of power may result from a mismatch at the joining of two dissimilar line sections.

(6) There may be power losses in pieces of equipment that are part of the communication circuit.

e. Power losses resulting from the characteristics of the line are reduced by loading, which increases the inductance in the line. The increased inductance increases the characteristic impedance of the line, which reduces the current in the line and the I^2R losses. Loading also causes the characteristic impedance and attenuation to be constant over a wide range of frequency.

f. Interference, which reduces the quality of the transmitted signal, is caused in the following ways:

(1) Electrostatic coupling exists between 2 pairs of lines when the electric field of 1 pair causes a voltage to be developed in the second pair. Such coupling results from an unbalanced condition between the pairs of conductors, or an unbalanced condition between the con ductors and ground.

(2) Electromagnetic coupling exists between 2 pairs when the magnetic field of 1 pair induces a voltage in the second pair. Such coupling may result from a differ ence in the distance between the 2 conductors of 1 pair and the wires of a second pair.

(3) Conductive coupling exists when there is leakage between two pairs of conductors as a result of faulty insulation between them.

(4) Coupling caused by a resistive unbalance exists when there is a difference in the resistances of the two wires of a pair. As a result, the currents induced in the two wires are unequal, even if the induced voltages are equal.

g. The effects of interference can be minimized by—

(1) Separating any pair carrying a weak current from any other pair carrying a strong current.

(2) Separating the receiving end of one pair from the sending ends of other pairs.

(3) Placing the conductors of 2 pairs so that the wires of 1 pair are at equal distances from the wires of the second pair.

(4) Spacing the conductors so that the distance between the wires of a pair is small, relative to the distance between pairs.

(5) Transposing the wires of a pair, which exchanges the positions of the two conductors of each pair at intervals along the line.

h. Interference on cables is principally due to electrostatic coupling. The interference is reduced by—

(1) Connecting small capacitors across the line.

(2) Splicing a section of cable having a low value of distributed capacitance to a section of cable having a high value of distributed capacitance.

i. The 4 conductors of 2 communication circuits may be used as the conductors for a third circuit, called a phantom circuit. The 2 wires of 1 side circuit act as a single conductor for the phantom circuit, and the 2 wires of the second side circuit act as the other conductor for the phantom circuit. To prevent interference between the side and phantom circuits, the phantom circuit must be balanced carefully.

37. Review Questions

a. Name the four distributed constants of a transmission line. Give the symbol for each constant, the units in which each constant is expressed, and the characteristics of the transmission line on which each of the distributed constants depends.

b. Define characteristic impedance. Which characteristics of the transmission line determine the characteristic impedance? State the relationship between the characteristic impedance of a transmission line and the maximum transfer of power to a receiver at the end of the line.

c. Which of the distributed constants of a transmission line determines the amount of attenuation? What produces frequency distortion? What is the effect of frequency distortion?

d. Define phase velocity. Define group velocity. Under what conditions are the two velocities equal? Unequal?

e. Define the phase-shift constant. What produces phase distortion? What is the effect of phase distortion?

f. Define attenuation. What produces attenuation? What produces frequency distortion? What is the effect of frequency distortion?

g. What are the usual causes of reflection? What are the effects of reflection?

h. What are some of the physical characteristics of an open-wire line?

i. What is a quad? What is a multiple-twin quad? What is a spiral-four quad?

j. How do lead-covered cables differ from rubber-covered cables? Where can lead-covered cables be used? Where can rubber-covered cables be used? Where can submarine cables be used?

k. What are some of the physical characteristics of field wires? Where can field wires be used?

l. Define loading. State its purpose. State its effect. What are the two methods of accomplishing loading?

m. Define balance to ground. How can an unbalance be produced? What could be the effect of an unbalance to ground? What is the principal source of interference coupled into a transmission line by an unbalance to ground?

n. Define unbalance between pairs. How is such an unbalance produced?

o. What are the three methods of coupling between pairs of transmission lines that cause interference?

p. Why does the separation between high power-level lines and low power-level lines reduce interference?

q. What type of configuration of the 4 wires of 2 pairs of conductors reduces interference produced by electromagnetic coupling? Why is a reduction of interference produced?

r. How does the proper spacing between the 2 wires of a pair, and between 2 pairs, reduce interference?

s. How is interference coupling reduced on long lines using repeaters?

t. What are transpositions? How do transpositions reduce interference? Why are the conductors transposed at frequent intervals? How does reflection on the line reduce the effectiveness of the transposition of conductors?

u. Why is crosstalk stronger on an open-wire line than on a cable? Why does electromagnetic coupling produce a negligible amount of crosstalk in a cable? What two methods are used to reduce interference by electrostatic coupling in a cable?

v. How is interference reduced in cables carrying carrier frequencies?

w. Define a phantom circuit. How can interference be prevented between a phantom circuit and the associated side circuits?

CHAPTER 3
VOICE-FREQUENCY REPEATERS

38. Introduction

a. Transmission-Line Problems. The primary purpose of a communication transmission line is to transmit a signal of sufficient power to operate properly the equipment at the receiving end of the line. In a telephone communication system, this means that the signal at the receiving end must be received with sufficient power to make the sound output of the telephone receiver audible above normal surrounding noise. A transmission line always attenuates the signal. Further, it may introduce distortion or interference that makes the output message unintelligible, even though it is audible. Both of these effects become increasingly important with increase in the length of the line.

b. Need for Amplification.

(1) This attenuation of the signal would limit the spacing between switchboards to relatively short distances, if the power lost on the line could not be replaced. The power can be, and is, replaced by the use of *amplifiers*—vacuum-tube devices which amplify the signal, deriving the additional power from d-c (direct-current) power sources.

(2) The triangle shown in A, figure 52, is the *symbol* for an amplifier and its circuit. Note that the point of the triangle points in the direction of the output. B shows in detail a typical *amplifier circuit* represented by the symbol. When a single vacuum-tube amplifier does not give a large enough gain, a multistage amplifier may be used.

c. Use of Amplifiers.

(1) The lowest permissible power level at a telephone receiver is approximately −30 dbm (decibels referred to 1 milliwatt in 600 ohms). Refer to chapter 6 for a discussion of dbm and db. A signal below this level will not produce a sound that is audible above average room noise. The one-way transmission line shown in A, figure 53, without amplifiers, shows a power drop to −40 dbm, which is below the permissible level. B, the same transmission line with the addition of an amplifier at each switchboard and one intermediate amplifier, shows a power output of −6 dbm, the preferred level for military use.

(2) The power-level diagram in A shows a straight-line drop to −40 *dbm*. In B, the diagram shows the drops in the line between the amplifiers, the gains through the amplifiers, and the desired −6 *dbm* power level at switchboard No. 2.

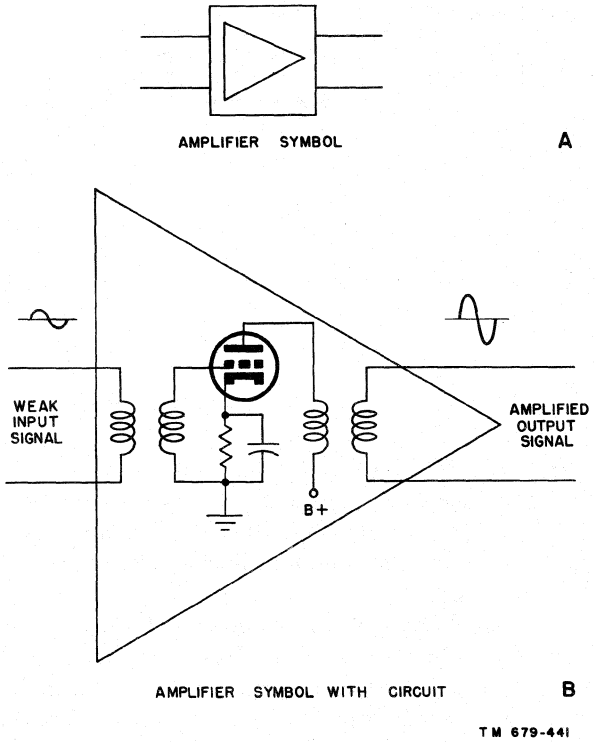


Figure 52. Amplifier symbol and circuit.

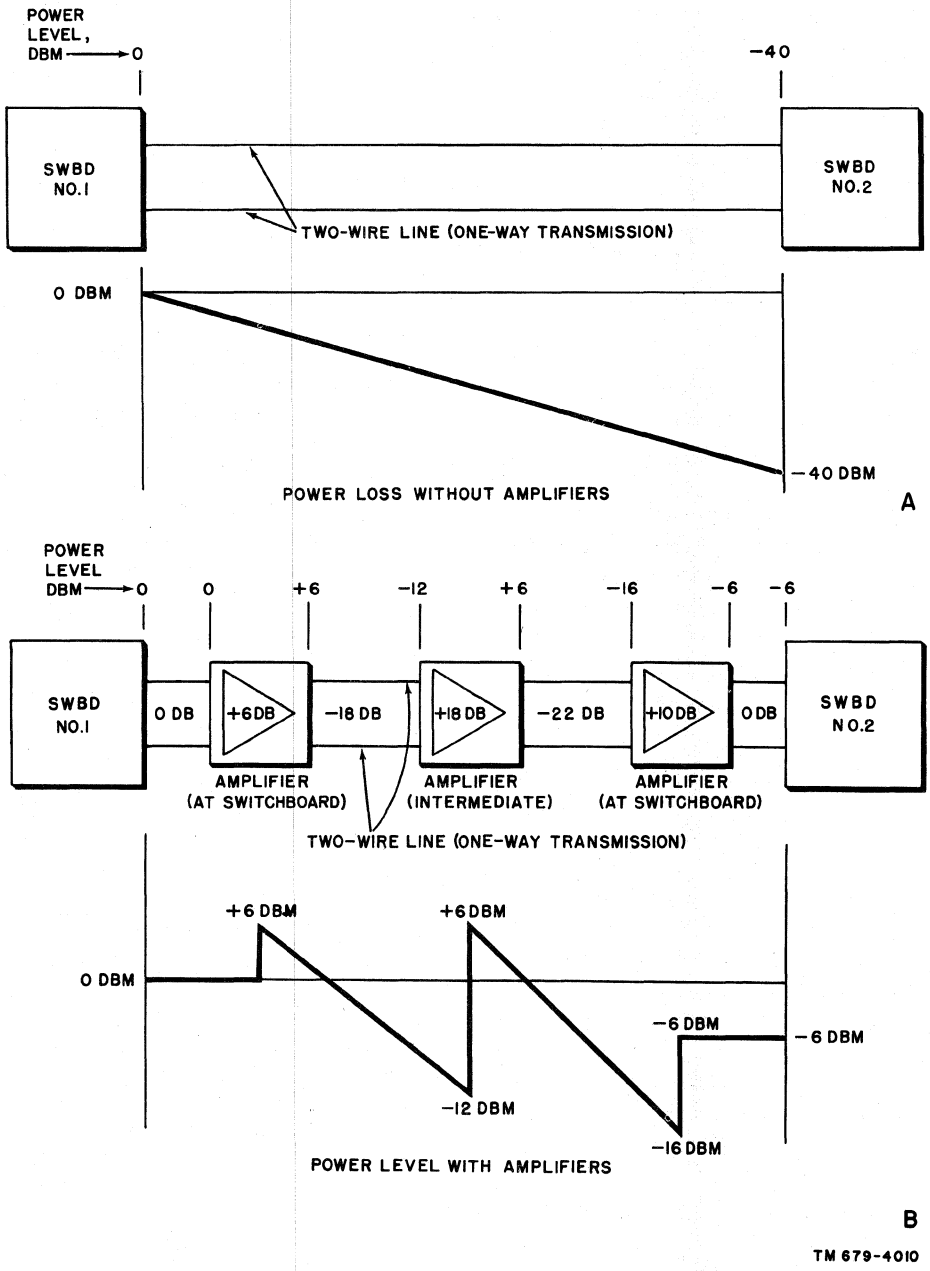


Figure 53. Power levels with and without amplifiers.

d. Need for Hybrid Arrangements.

(1) Telephony requires two-way transmission. This cannot be effected with amplifiers alone, for the amplifier is a one-way device.

(*a*) Figure 54 shows two amplifiers in a two-wire line. This circuit will not work. The output signal of either amplifier feeds into the farther line, as desired, but it also feeds into the input of the other amplifier; then, emerging reamplified, it feeds back through the line it came from and again feeds through the first amplifier, and so on, around and around. A complete *feedback loop* is made, and random oscillations occur. A wide range of frequencies is generated, including frequencies in the voice range. The voice energy circulating through the two amplifiers is amplified

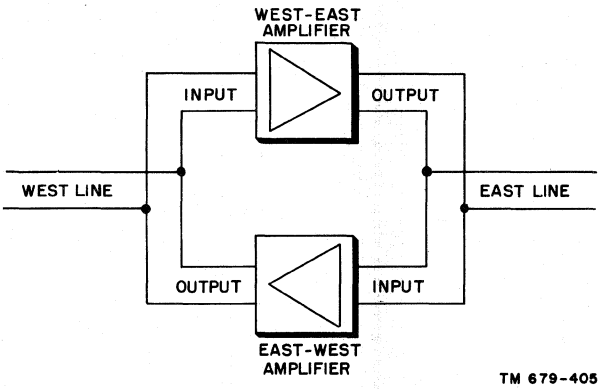


Figure 54. Repeater circuit that produces singing.

until the repeater *howls* or *sings*. In this condition, the repeater is of no use whatsoever.

(*b*) For example, assume that each amplifier of a system has a gain of +20 *db* and an output of +6 *dbm* when in the line singly. When connected, as in figure 54, an incoming signal from the east linel eaves the E–W amplifier at +6 *dbm*. The west line will take half the signal power (approximately). A +3-*dbm* signal, therefore, is sent out on the West line and a +3-*dbm* signal is fed into the W–E amplifier. The signal level is raised to +23 *dbm* at the output of the W–E amplifier, and then divides equally (approximately) between the East line and the input of the E–W amplifier, each path this time receiving a signal at a +11.5-*dbm* level, and so on. This building up of power will not continue indefinitely, however. When the input signal becomes too great, overloading of the amplifier occurs. The overall results of this are distortion and singing, which impair operation to the point where transmission is impossible. Reduction of the amplifier gains would reduce the singing, but it also would reduce the signal power in the outgoing line, thus defeating the purpose of the amplifier. Two amplifiers by themselves, then, cannot be used in a two-wire line.

(2) In figure 53, B shows a two-wire system that can be used for *one-way transmission*. Doubling this system, by adding alongside a similar system with the amplifiers operating in the opposite direction, would give a workable four-wire system; but, for connection through each switchboard, means would have to be provided to bring the four wires of the line to the two terminating wires required at the switchboard.

(3) Two amplifiers *can* be used for 2-way transmission in a 2-wire system, *and* the 4 wires of a 4-wire system can be terminated into 2 wires for use through a switchboard, by the addition of certain *coils* used in *special networks* called *hybrid arrangements*, or *hybrid sets*. These will be discussed shortly.

39. Repeater Systems

The combination of an amplifier or amplifiers with attached networks, such as hybrid arrangements, is called a *repeater*. Repeaters used at the ends of a transmission line are called *terminal repeaters;* those used between the ends of a transmission line are called *intermediate repeaters*. A system may have a number of intermediate repeaters, or none at all.

a. Two-Wire Repeater System.

(1) A, figure 55, illustrates how the amplifier and hybrid arrangements are connected in a repeater to permit two-way transmission in the important *two-wire system*. The system consists of two switchboards, two terminal repeaters, one intermediate repeater, and a two-wire transmission line. Each repeater contains two amplifiers which transmit in opposite directions, the upper amplifiers transmitting from West to East, the lower amplifiers from East to West. The uppermost row of figures indicates the power level in dbm at the inputs and outputs of the switchboards and repeaters for transmission from West to East; the lowest row indicates the same for transmission from East to West. The figures next to the amplifiers indicate their gains of the amplifiers. The figures within the sections of line indicate the losses in the sections of line.

(2) The power level of a signal leaving switchboard No. 1 is 0 *dbm* (A, fig. 55). The power level at the input of the adjoining terminal repeater also is 0 *dbm* (for all practical purposes, because the length of

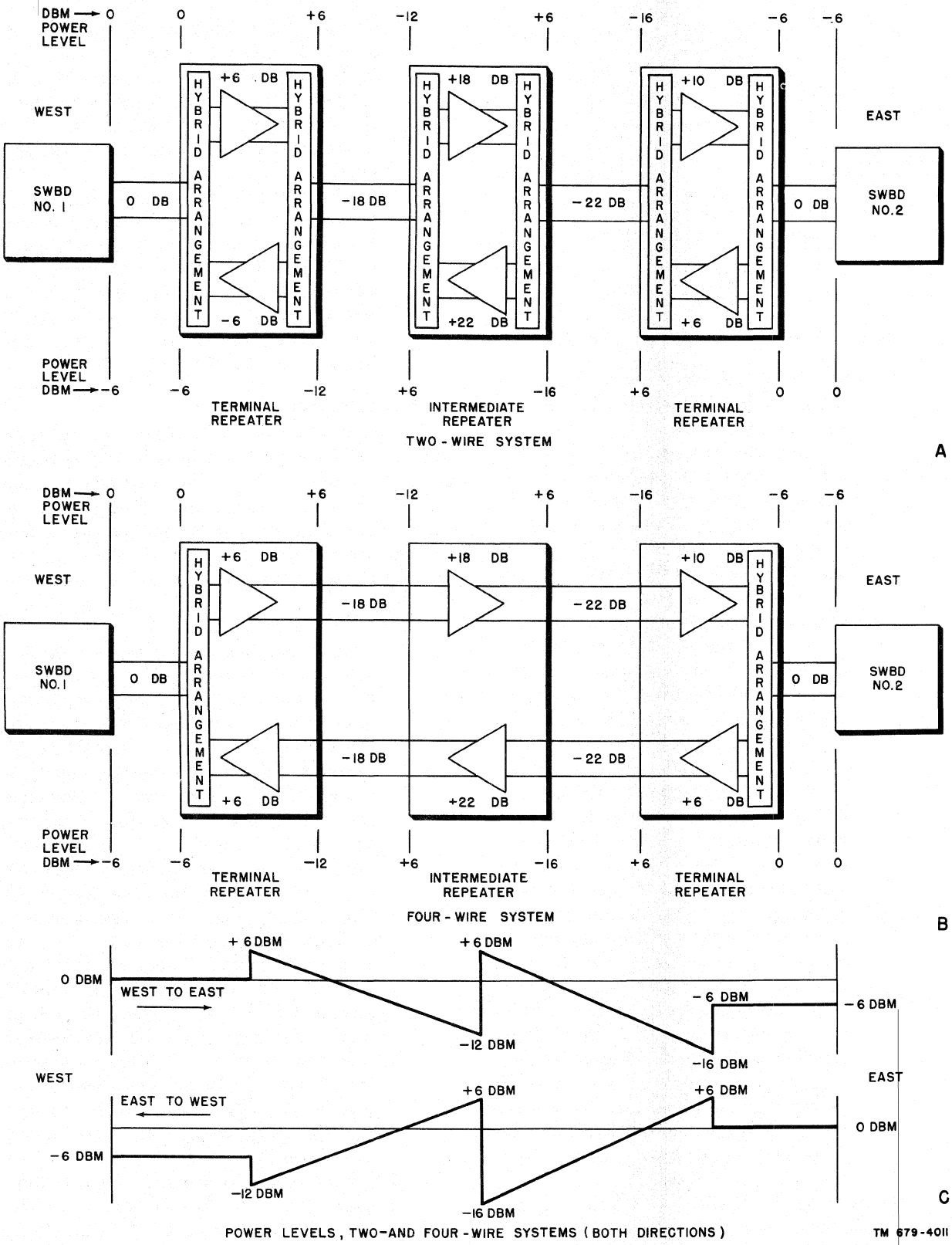


Figure 55. Two- and four-wire systems.

the intervening line section is so short that the loss is negligible). This 0-*db* loss in the line section is indicated between the wires; both of the 0-*dbm* levels are indicated in the top row of figures. The net gain of the terminal repeater in this direction of transmission is 6 *db*, as indicated by the figure above the amplifier. This gain raises the level of the signal power at the output of the repeater to +6 *dbm*, as indicated above the repeater. The next line segment causes a loss of 18 *db*, as indicated within the wires of the line section. This loss reduces the level of the signal power at the input to the intermediate repeater to −12 *dbm*, as indicated. The net gain of the intermediate repeater in this direction of transmission is 18 *db*. This gain raises the level of the signal power to +6 *dbm* at the output of the repeater.

(3) The process continues in this manner, the upper amplifiers of the repeaters making up all or part of the losses of the sections of line just to their left (A, fig. 55). The gains are so arranged that the level of the signal power at the input to switchboard No. 2 is the desired −6 *dbm*. For the opposite direction of transmission the process is the same, losses and gains in *db*, and power levels in *dbm*, as indicated by the lower set of values. Each amplifier has a separate gain control. Note that the amplification of a repeater is not necessarily the same in both directions.

(4) The power level at each point in the system for both directions of transmission is shown in C, figure 55.

b. Four-Wire Repeater System.

(1) In contrast with A in figure 55, B shows how the amplifiers and hybrid arrangements are connected in the repeaters of the equally important *four-wire system*. The values expressed by the numbers stand for the same things as those given in the two-wire system represented above. For purposes of comparison, the two sets of values have been made identical. Note that one side of each terminal repeater is connected through a *hybrid arrangement* to the 4 wires of the transmission line, and that the other side is connected to the 2 wires leading into the switchboards. The intermediate repeater uses only the 2 amplifiers (with associated equipment but without hybrid arrangements), 1 for each direction of transmission.

(2) C, figure 55, shows the power level at each point in the system for both directions of transmissions.

(3) Figure 56 is the front view of a typical repeater. The unit contains two single-stage amplifiers which operate in opposite directions. Note that each amplifier has its own fine and coarse gain controls. Figure 57, a rear view, shows the orderly arrangement of the many wires and terminals. The unit is approximately 20 inches wide.

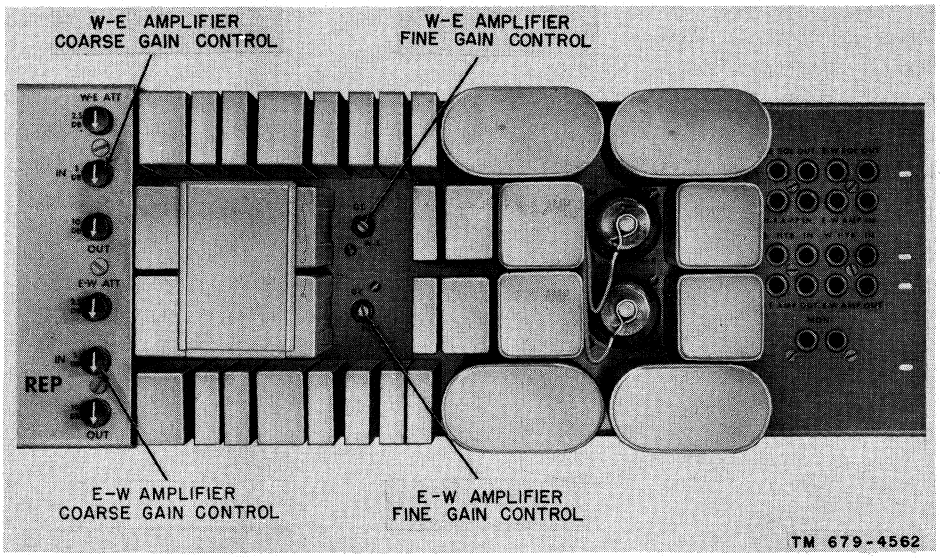


Figure 56. Repeater panel, front view.

282463 O—54——4

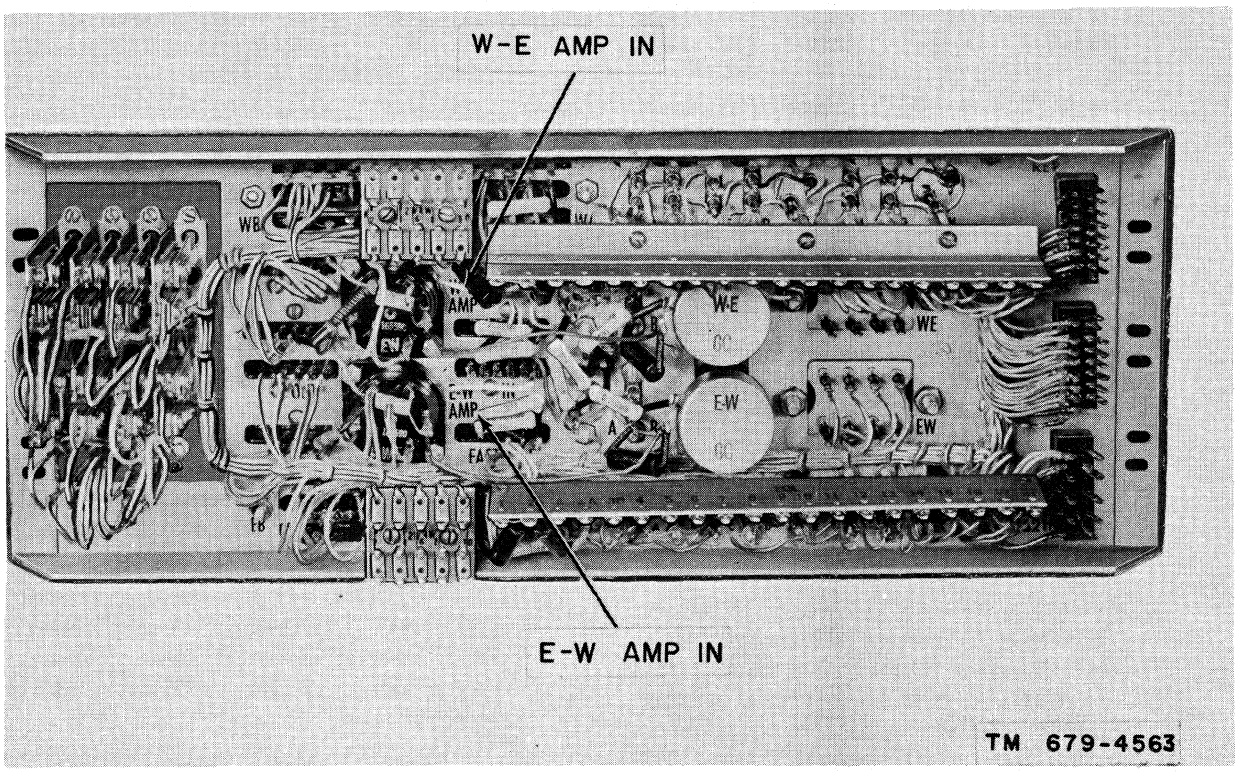


Figure 57. Repeater panel, rear view.

40. Basic Principles of Hybrid Arrangements

a. Types of Hybrid Arrangements. The four types of hybrid arrangements are *the hybrid-coil arrangement*, which employs the hybrid coil; *the repeating-coil hybrid arrangement*, which employs the repeating-coil; *the four-wire terminating set*, which employs the four-winding coil; and *the resistance hybrid*, which employs resistances only.

b. Underlying Principles of Hybrid Arrangements. Underlying the operation of hybrid arrangements is one or more of the principles of the *direct-current Wheatstone bridge*, the *alternating-current Wheatstone bridge*, and *transformer coupling*.

c. Principle Underlying Each Hybrid Type.

(1) The *hybrid-coil arrangement* embodies the principle of the a-c Wheatstone bridge and transformer coupling. It is a transformer device.

(2) The *repeating-coil hybrid arrangement* embodies the principle of the transformer. It is a transformer device.

(3) The *four-wire terminating* set embodies the principle of the transformer. It is a transformer device.

(4) The *resistance hybrid* embodies the principle of the d-c Wheatstone bridge. It is a resistance device.

d. Reference. For the three principles underlying the four hybrid arrangements, see paragraphs 41–43. See also TM 11–661 and TM 11–681.

41. Principle of Direct-Current Wheatstone Bridge

a. Figure 58 shows a *d-c Wheatstone bridge*, with a battery connected across points A and B and a galvanometer across C and D. The battery causes current to flow in resistance arms $R1$, $R2$, $R3$, and $R4$. Because in a series circuit the voltages across the resistances are proportional to the

resistances, $R1$ can be—and in this case is assumed to be—of such value that

$$\frac{R1}{R2}=\frac{R3}{R4}.$$

The voltage drop across $R1$, therefore, is equal to the voltage drop across $R3$. This being so, points C and D are at the same potential and the galvanometer shows no current flow between them. The bridge is said to be *balanced,* or *in balance.* Assume that $R1$ is set to any other value. $R1/R2$ now does not equal $R3/R4$; a potential difference exists across points C and D, and a current flows in the galvanometer, indicating that the bridge is no longer in balance.

b. Consider this example: Find the value of $R1$ at balance when $R2=150$ ohms, $R3=300$ ohms, and $R4=900$ ohms. At balance, as stated previously,

$$\frac{R1}{R2}=\frac{R3}{R4},$$

or

$$R1=R2\,\frac{R3}{R4}.$$

Substitute the given values of $R2$, $R3$, and $R4$, and obtain

$$R1=150\,\frac{300}{900}=50 \text{ ohms}.$$

The simplest condition for balance is obtained when all the bridge arms are set equal to each other.

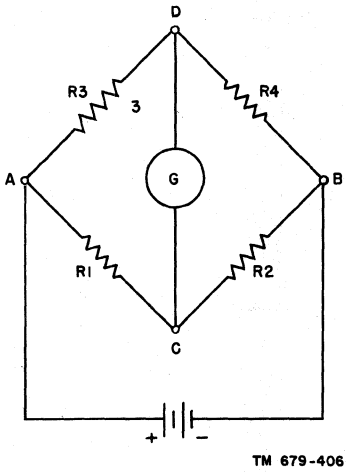


Figure 58. Direct-current Wheatstone bridge.

42. Principle of Alternating-Current Wheatstone Bridge

Figure 59 shows an *a-c Wheatstone bridge,* also called an *impedance bridge.* Like the d-c bridge arrangement, the impedance bridge has four arms, but it has an a-c generator (voltage source) across points A and B, and a headset (a-c detecting device) across points C and D. The a-c generator operates at some voice frequency (1,000 cycles per second is a commonly used value). Impedance $Z1$, $Z2$, $Z3$, and $Z4$ are the bridge arms. The generator causes current to flow through all the impedances of the bridge arms. Because in a series of circuit the voltages across the impedances are proportional to the impedances, Z1 can be—and in this case is assumed to be—of such value that

$$\frac{Z1}{Z2}=\frac{Z3}{Z4}.$$

The voltage drop across $Z1$ is therefore equal to the voltage drop across $Z3$. This being so, points C and D are at the same potential, and there is no current flow between them (no sound from the headset). The bridge is in balance. Assume that $Z1$ is set to any other value. $Z1/Z2$ now does not equal $Z3/Z4$, a potential difference exists across points C and D, and an alternating current flows in the headset; this causes sound to be emitted from the headset, indicating that the bridge is no longer in balance. The greater the sound, the greater is the bridge unbalance. Note that alternating current flows in the bridge arms at balance.

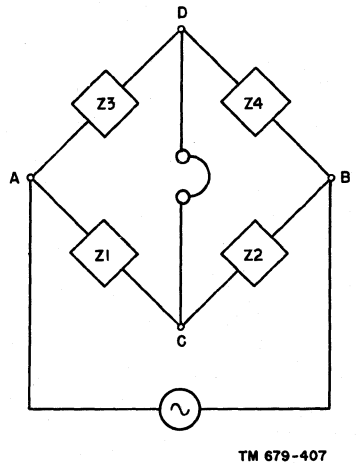


Figure 59. Alternating-current Wheatstone bridge.

43. Principle of Transformer Coupling

a. Transformer coupling is used for impedance matching and the electrical isolation of circuits. Figure 60 shows a *transformer-coupled arrangement.* The transformer couples inductively the a-c generator to the load impedance $Z2$. The internal impedance of the generator, Z_g, is shown in series with the generator. The impedance looking into the primary is $Z_1=E_1/I_1$ and the secondary load is $Z_2=E_2/I_2$. In an ideal transformer,

$$\text{(A)} \qquad \frac{N_1}{N_2}=\frac{E_1}{E_2}=\frac{I_2}{I_1}$$

where N_1 and N_2 are the number of turns of the primary and secondary winding, respectively. From equation (A),

$$\text{(B)} \qquad E_1=\frac{N_1}{N_2}E_2 \text{ and}$$

$$\text{(C)} \qquad I_1=\frac{N_2}{N_1}I_2.$$

Dividing equation (B) by equation (C),

$$\text{(D)} \qquad \frac{E_1}{I_1}=\left[\frac{N_1}{N_2}\right]^2\frac{E_2}{I_2}.$$

Substituting in equation (D),

$$\frac{E_1}{I_1}=Z1 \text{ and } \frac{E_2}{I_2}=Z2.$$

results in

$$\text{(E)} \qquad Z1=\left[\frac{N_1}{N_2}\right]^2 Z2.$$

In order to obtain maximum power transfer from the generator to the load, $Z1$ is set equal to the internal impedance of the generator (Z_g) in equation (E). The transformer then is said to match the load to the generator. Maximum power transfer from a generator of fixed internal impedance is obtained by setting the load impedance equal to it. In this case, maximum power is transferred to the primary winding of the transformer; and, since an ideal transformer does not dissipate any power, all of the power delivered to the transformer goes to the load, $Z2$. Setting $Z1$ equal to Z_g in equation (E) gives

$$Z_g=\left[\frac{N_1}{N_2}\right]^2 Z2.$$

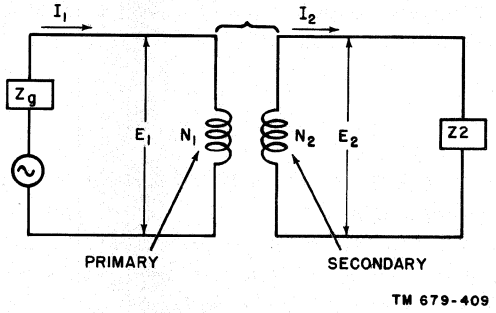


Figure 60. Transformer coupling.

b. Consider this numerical example: Find the turns ratio required if Z_g is 10,000 ohms and Z_2 is 100 ohms.

$$\text{(F)} \qquad Z_g=\left[\frac{N_1}{N_2}\right]^2 Z2$$

or

$$\frac{N_1}{N_2}=\sqrt{\frac{Z_g}{Z_2}}=\sqrt{\frac{10{,}000}{100}}=\sqrt{100}$$

or

$$\frac{N_1}{N_2}=\frac{10}{1}.$$

If $N_1=100$, $N_2=10$. It is clear that matching depends upon the turns ratio.

44. Impedance Bridge Used as Hybrid Arrangement

Paragraph 45 discusses the hybrid-coil arrangement, first of the four types of hybrid arrangements. Here, as a helpful transitional step, a modified impedance bridge (fig. 61) will be considered as a hybrid arrangement.

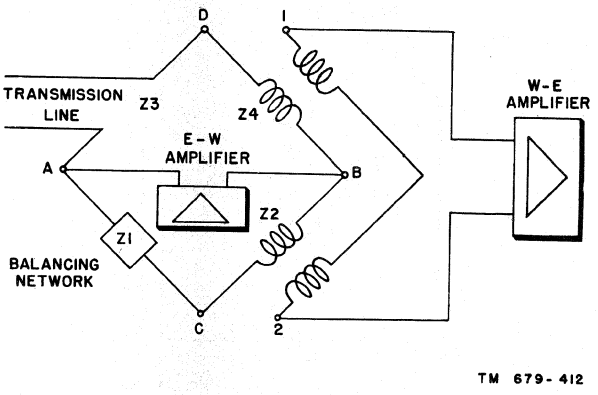


Figure 61. Modified impedance bridge used as a hybrid arrangement.

a. Description of Arrangement. Figure 61 is obtained by modifying the a-c Wheatstone-bridge circuit. $Z3$ of figure 59 is replaced by a two-wire transmission line; the a-c generator is replaced by the output of the E–W amplifier; the headset is in effect replaced by the input to the W–E amplifier, since $Z2$ and $Z4$ are now transformer primary windings C–B and B–D respectively. Actually, C–D is a single winding that is center-tapped at point B. Winding 1–2 is the transformer secondary and is connected to the W–E amplifier input. Impedance $Z1$ is adjustable; it is made to equal $Z3$, the transmission-line impedance. Since $Z2$ equals $Z4$, this satisfies the balance requirement that $Z1/Z2=Z3/Z4$. Impedance $Z1$ is called the *balancing impedance* or, more commonly, the *balancing network*. This arrangement has no feedback loop; the output of the E–W amplifier does not reach the input of the W–E amplifier and does reach the transmission line. This is shown clearly by an investigation of the East–West (E–W) and the West–East (W–E) paths of the signals.

b. East–West Path. The E–W path is the path taken by useful signal in going from the E–W amplifier to the two-wire transmission line (fig. 61). The E–W path starts at the output of the E–W amplifier. The E–W amplifier output is across A–B. Since $Z2$ equals $Z4$ and $Z1$ equals $Z3$, the current in the B–D winding (fig. 61) is equal to the current in the B–C winding when a signal is across A–B. The currents induce equal voltages of opposite polarity into the 1–2 winding, however. These voltages cancel each other, and no signal reaches the W–E amplifier input. Thus the signal from the output of the E–W amplifier does not reach the input of the other amplifier. There is no feedback loop, and no singing. A potential difference exists across A–D, however, and signal is sent out into the line. $Z1$, the balancing network, dissipates as much power as is sent into the transmission line, since it is equal to the line impedance and has the same potential difference across it.

c. West–East Path. The W–E path is the path that useful signal takes in going from the transmission line to the W–E amplifier input (fig. 61). The W–E path starts at the transmission-line terminals A and D. The circuit is not in balance for the incoming signal. Potential differences are developed across A–B and C–D. The signal across A–B is completely dissipated in the E–W amplifier and is not amplified, since the amplifier is a one-way device. The potential difference across primary winding C–D causes current to flow and induce voltage in winding 1–2, and signal reaches the W–E amplifier input. The power level of the signal will be raised, for further transmission, as it passes through the W–E amplifier.

d. Need for Modification of Arrangement. Other factors require a physical modification of this circuit, but the underlying principles remain the same.

45. Hybrid-Coil Arrangement

Figure 61 is not drawn in the conventional way that is used to represent the hybrid-coil arrangement; it is intended to stress the balanced-bridge principle involved in this particular type. Figure 62 (except for one change) is the conventional diagram. All point designations have been kept, to facilitate comparison with figure 61. Figure 62, like figure 61, contains an unbalanced-line condition which must be pointed out before the *workable* hybrid coil is discussed.

a. Unbalanced-Line Hybrid Arrangement. In figure 62, one side of the transmission line is terminated at a winding and the other side is connected to the balancing network. The line is not terminated in a symmetrical network. The result is an unbalanced line and the introduction of crosstalk (interfering conversations or transmissions from other lines). This condition is not desired and, in practice, the arrangement is not used.

b. Balanced-Line Hybrid Arrangement. Figure 63 shows the *hybrid coil* in a *hybrid-coil arrangement* used extensively in practice. The arrangement is like the former one, but the coil differs in that half of winding B-C (fig. 62) is placed between the balancing network and point A (A–F′ in fig. 63) and half of winding D–B (fig. 62) is placed on

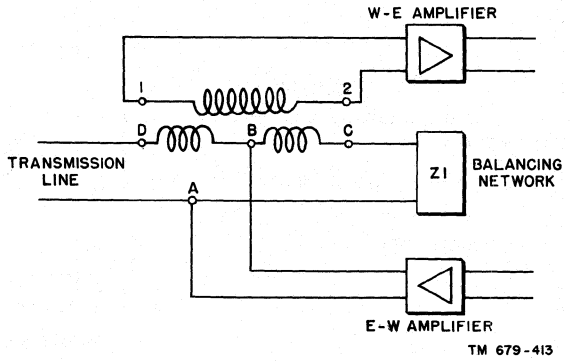


Figure 62. Unbalanced-line hybrid arrangement.

the line side of point A (E′–A in fig. 63). This is done so that perfect symmetry in the wiring of the voice circuit will exist. Winding voltage polarities are indicated by plus and minus signs. The transmission line terminates in a symmetrical network, and the result is a balanced line. In the coil itself, the two windings between C′ and D′ and between E′ and F′ are single windings with center taps at points B and A respectively. All three windings are wound on the same magnetic core. They are inductively coupled and form one transformer. This hybrid coil, with its three inductively coupled windings, can efficiently separate the E–W and W–E signal paths at the *junction of a two-wire and a four-wire system* when used in the arrangement shown.

c. Operation of Hybrid-Coil Arrangement.

(1) *Operation of hybrid-coil arrangement in transmission from E–W amplifier to line.* In the hybrid-coil arrangement, the output signal from the E–W amplifier is across A–B (fig. 63). Looking from A–B, the series networks toward the transmission line and toward the balancing network are symmetrical; that is, windings BD′ and BC′ are identical, windings AE′ and AF′ are identical, and the impedances at the line and balancing network are equal. Therefore, the current that flows from point B through the transmission line back to point A is equal to the current that flows from point B through the balancing network back to point A. Because the currents flow in opposite directions in windings BD′ and BC′ and in windings AE′ and AF′, equal and opposite voltages are induced into winding 1–2. Being equal and opposite, the voltages cancel completely in winding 1–2; no signal reaches the input of the W–E amplifier; there is no feedback loop and no singing; and signal power is sent out into the transmission line and to the balancing network. And, since the transmission line and balancing-network impedances are equal and receive equal currents, the power sent out into the line is approximately equal to the power dissipated in the balancing network. Thus, only about 50 percent of the power from the E–W amplifier actually is transmitted to the transmission line. This loss of power amounts to a 3.25-decibel loss.

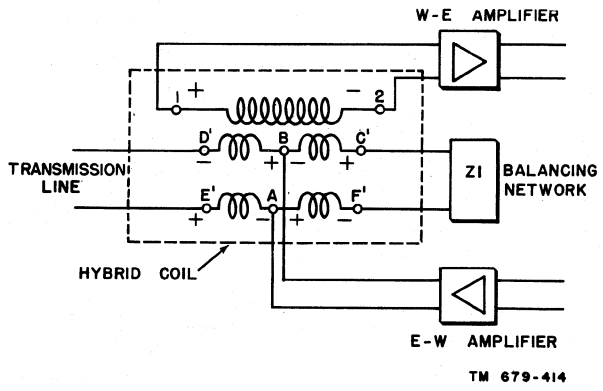


Figure 63. Hybrid-coil arrangement.

(2) *Operation of hybrid-coil arrangement in transmission from line to W–E amplifier.* When a signal voltage (fig. 63) is impressed across the transmission line, current flows through the windings AE′, through the plate circuit of the amplifier tube in the E–W amplifier, and back to the other side of the transmission line through winding BD′. The current flow through winding AE′ induces a voltage in winding 1–2. This induced voltage is impressed on the input grid of the amplifier tube in the W–E amplifier. However, only one-half of the power from the transmission line actually reaches the W–E amplifier; the rest of the power is lost in the plate circuit of the amplifier tube in the E–W amplifier. This represents a 3.25-decibel loss. Furthermore, no signal power reaches the balancing network. To understand why this is true, refer to figure 64. This figure is equivalent to figure 62, differing only in that the line is replaced by a generator and its equivalent impedance Z_L, and the plate circuit of the E–W amplifier is represented by the impedance Z_A. Z'_A is the input impedance of the W–E amplifier. Windings DB and BC have the same number of turns, are wound on the same core, and are inductively coupled. Therefore, current flow in winding DB induces a voltage in winding BC. The circuit is designed so that the voltage induced in winding BC is exactly equal to the voltage drop across Z_A (the plate circuit of the E–W amplifier).

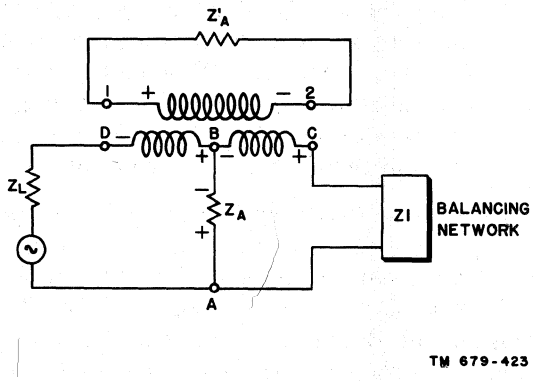


Figure 64. Analysis of balancing-network loop.

Consequently, no difference in potential exists across the balancing network and there can be no current flow between these two points of equal potential.

d. Arrangement of Hybrid Coils in Two-Wire Repeater. Figure 65 shows how two hybrid coils are connected to the amplifiers in a repeater. This repeater is used in a *two-wire repeater system.*

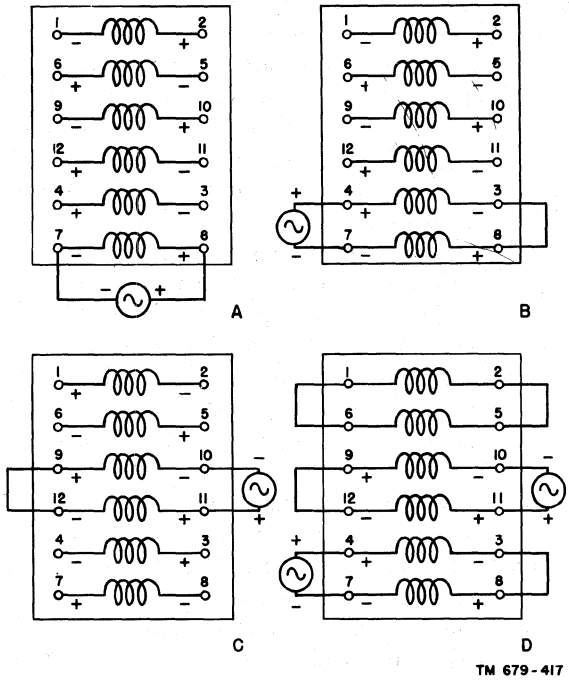


Figure 66. Repeating-coil, 173-type.

46. Repeating-Coil Hybrid Arrangement

a. Description of Repeating Coil. A repeating coil is a transformer arrangement consisting of two or more windings. Its function is to *repeat* the energy from one circuit into another circuit, rather than to change the voltage and current values. A, figure 66, is a schematic diagram of a *173-type repeating coil.* The coil consists of six separate windings, each of which is magnetically coupled to all the others. Assume that the generator gives winding 7–8 the polarity shown. Voltages are induced in all the other windings, and they have the instantaneous polarity shown. The rule to be followed when considering the polarity of the induced voltages in these windings is: *Odd-numbered terminals have the same polarity, and even-numbered terminals have the same polarity.* The odd-numbered and even-numbered terminals have opposite polarities. When the generator polarity reverses, the polarity of the induced voltages in all windings also reverses.

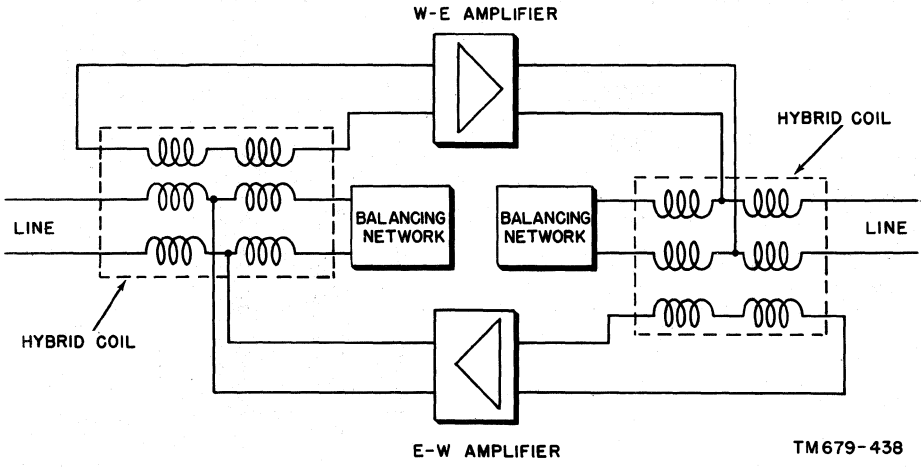


Figure 65. Arrangement of hybrid coils in two-wire repeater.

b. Principle of Repeating Coil.

(1) Place the generator across terminals 4–7 and connect terminals 3–8 (B, fig. 66). The positive side of the generator is still connected to an even-numbered terminal and the negative side to an odd-numbered terminal; therefore, there is no change of voltage polarity in the two figures. Now, instead, place the generator across terminals 10–11 and connect terminals 9–12, as in C. The positive side of the generator now is connected to an odd-numbered terminal and the negative side to an even-numbered terminal, and there is a change of voltage polarity through the coil, as compared to B. Now, note that in D there is no voltage across windings 1–2 and 5–6.

(2) The explanation of the latter condition involves the *principle of opposing induced voltages;* that is, opposing magnetic fields induce opposing voltages in a winding or pair of windings. When the magnitudes of the opposing magnetic fields are equal, the opposing voltages induced by the fields are equal, and the voltages cancel out. The strength of a magnetic field depends on the product of the number of turns of the winding and the current in the winding. Hence, for two magnetic fields to be equal in magnitude, the product of the number of turns and currents, respectively, must be equal. This situation is illustrated in D. Windings 7–8, 3–4, 11–12, and 9–10 all have the same number of turns. *Two* generators, identical, placed as in B and C, figure 66, are causing directly opposite magnetic fields in the windings to which they are connected. Windings 1–2 and 5–6 each have induced in them voltages of opposite polarity. The voltages cancel, and there is no flow of current.

c. Operation of Repeating-Coil Hybrid Arrangement. In figure 67, two repeating coils are used in a hybrid arrangement, and they are termed, for reference purposes, coil A and coil B. The windings of coil B have been rearranged to obtain a simpler diagram, but the coils themselves are physically identical. The two-wire line is balanced to ground since each line side is terminated in the same number of identical windings. The amplifiers and transmission line are magnetically coupled. The task of the repeating-coil hybrid arrangement is to give good transmission in both directions.

(1) *Operation of repeating-coil hybrid arrangement in transmission from E–W amplifier to line.* In figure 67, the *arrows* indicate *current directions;* the positive and negative *signs* indicate *voltage polarities.* Current from the output of the E–W amplifier flows in windings 2–1 and 6–5 of coil B, as indicated. Terminals 1 and 5 are positive, and terminals 2 and 6 are negative. Because of this, all the odd-numbered terminals of coil B are positive and all the even-numbered terminals are negative, simultaneously. Windings 9–10 and 11–12 of coil B apply their total induced series volatge to windings 9–10 and 11–12 of coil A. This voltage is applied in such a way

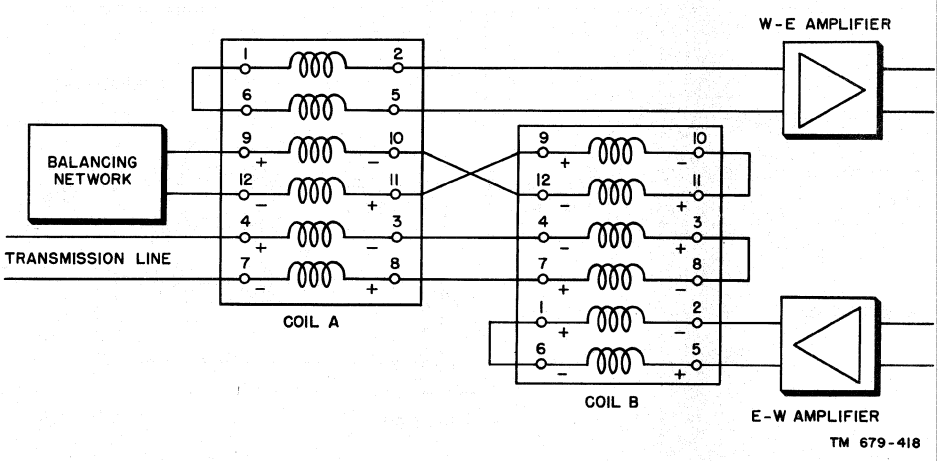


Figure 67. Repeating-coil hybrid arrangement in transmission from E–W amplifier to line.

that terminal 11 becomes positive and terminal 10 negative. In the same way, terminal 8 of coil A is positive, and terminal 3 of coil A is negative. In coil A, equal voltages are applied across terminals 10 and 11 and across terminals 3 and 8. Currents of equal magnitude then flow in the balancing network and line impedance. No net voltage is induced in windings 1–2 and 5–6; hence, no signal is received at the input to the W–E amplifier from the E–W amplifier, there is no feedback loop and no singing, and the West transmission line carries an outgoing signal.

(2) *Operation of repeating-coil hybrid arrangement in transmission from line to W–E amplifier.* The arrangement of the repeating-coil hybrid is such that a line signal finds its way both to the input of the W–E amplifier and the output of the E–W amplifier. The part that reaches the W–E amplifier is amplified and passed on, but the part that reaches the E–W amplifier is dissipated. The balancing network does not dissipate any of the incoming signal power. Figure 68 shows in detail what happens. The arrows indicate current directions and the signs indicate voltage polarities. Assume that the line signal causes current to flow in windings 7–8 and 3–4 of repeating coils A and B, as indicated. The line-signal current in windings 7–8 and 3–4 of coil A induces voltages into the other windings of coil A only. The same current in windings 7–8 and 3–4 of coil B induces voltages into the other windings of coil B. All the induced voltages have the indicated polarity. The total induced voltage in windings 9–10 and 11–12 of coil A equals the total induced voltage in the corresponding windings of coil B. All these windings are so connected that the total induced voltages in the balancing-network loop are subtractive, and cancel; hence, there is no current through the balancing network, and no power is delivered to it. The induced voltages in windings 1–2 and 5–6 of both coils are additive, however, and current flows. This places signal at both the input of the W–E amplifier and the output of the E–W amplifier. The results are an amplified signal at the W–E amplifier output and complete dissipation of signal that arrives at the output of the E–W amplifier.

d. Arrangement of Repeating-Coil Hybrids in Two-Wire Repeater. Figure 69 shows how two repeating-coil hybrids are connected to the amplifiers in a two-wire repeater. This repeater is used in a *two-wire repeater system.*

47. Four-Wire Terminating Set

Two simpler repeating coils—each containing four windings instead of six—can be used to obtain hybrid action, though they do not give as good performance as the hybrid coil or repeating-coil hybrid. A hybrid arrangement of two four-winding

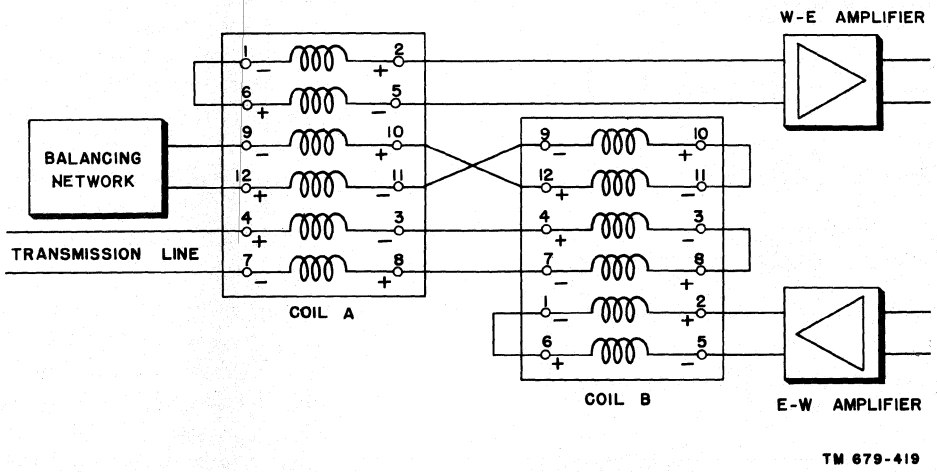


Figure 68. Repeating-coil hybrid arrangement in transmission from line to W–E amplifier.

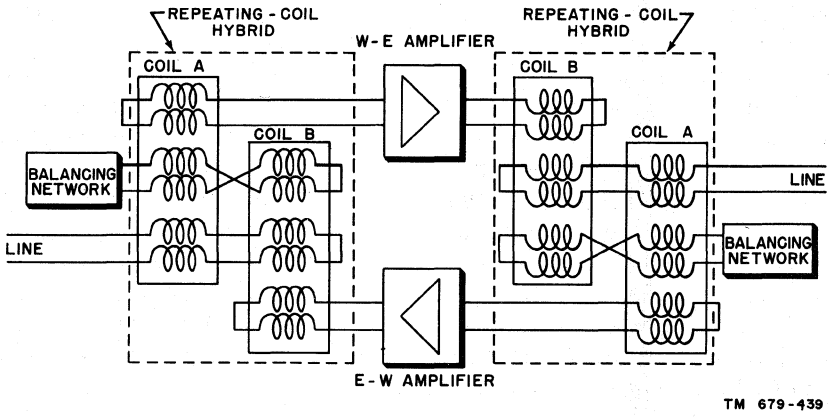


Figure 69. Arrangement of repeating-coil hybrids in two-wire repeater.

repeating coils is called a *four-wire terminating set.*

a. Four-Winding Repeating Coil. A *four-winding repeating coil* is represented in figure 70. Across terminals 7–8 is a generator, and the voltage polarities of the windings of the coil are as indicated. The signs indicate voltage polarities. The odd- and even-numbered terminals follow the voltage-polarity rule that all odd terminals have one polarity and all even terminals have the other polarity. The winding polarity determines the method of connecting the repeating coils, line, and amplifiers.

b. Operation of Four-Wire Terminating Set. The method of using two four-winding repeating coils in a four-wire terminating set is shown in figure 71. The arrows indicate current directions. For reference, the two coils used in this hybrid arrangement are termed coil A and coil B. The line is balanced to ground, since each line side is terminated in the same number of identical windings, and the amplifiers and line are coupled magnetically.

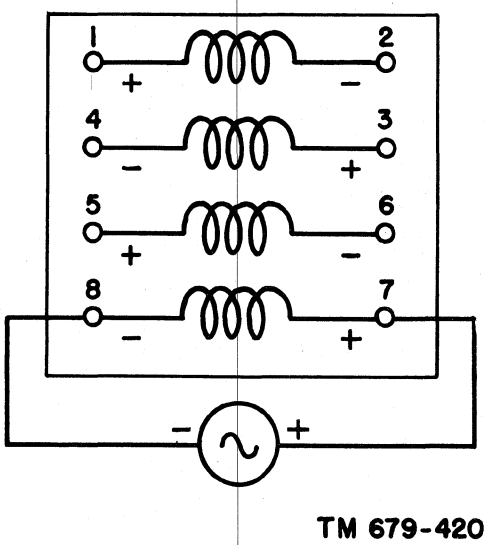


Figure 70. Four-winding repeating coil.

(1) *Operation of four-wire terminating set in transmission from E–W amplifier to line.* The output signal voltage from the E–W amplifier (fig. 71) causes the same current to flow in the 7–8 windings of both coil A and coil B. The two windings are so connected that they have opposite polarity. The induced voltage in winding 1–2 of coil A, therefore, is opposite to, and cancels, the induced voltage in winding 1–2 of coil B. This cancellation prevents the application of any signal to the input of the W–E amplifier. There is no feedback loop, and no singing. Simultaneously, the induced voltages in windings 3–4 and 5–6 of both coils are additive, and send currents to the line and balancing network. The currents are equal because the balancing network and line impedances are equal. If the currents in windings 3–4 and 5–6 of both coils were not equal, the voltages induced in the 1–2 windings would not be equal; signal power would reach the W–E amplifier input, and there would be a feedback loop and a probability of singing.

(2) *Operation of four-wire terminating set in transmission from line to W–E amplifier.* In figure 72, assume that the incoming line signal from the West sets up the indicated current directions and voltage polarities in windings 3–4 and 5–6 of coil A. This line current induces volt-

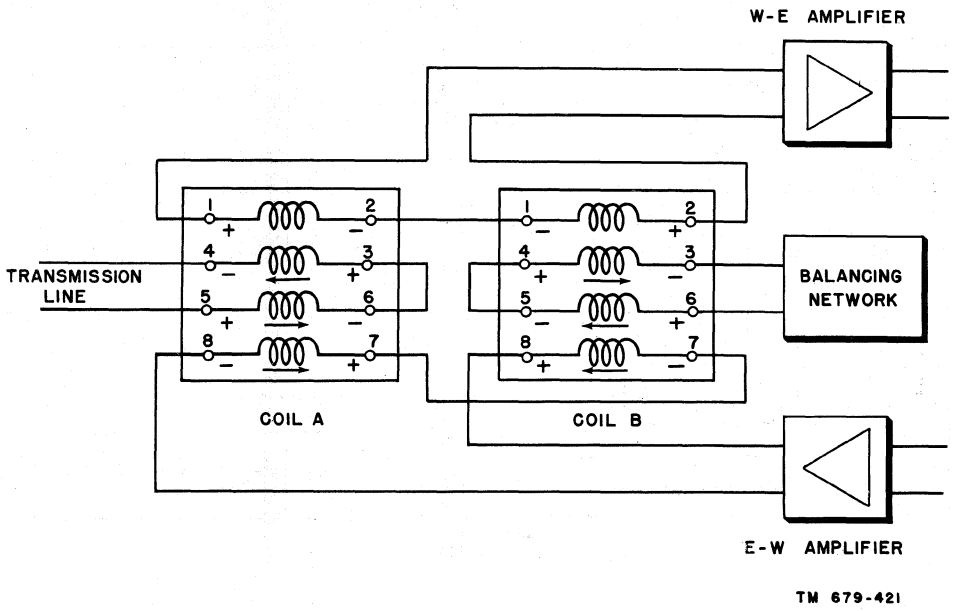


Figure 71. Four-wire terminating set in transmission from E–W amplifier to line.

ages in the 1–2 and 7–8 windings of coil A. The currents that flow in these windings also flow in the 1–2 and 7–8 windings of coil B. There they simultaneously induce voltages in the 3–4 and 5–6 windings of coil B. These windings are so connected that the induced voltages buck each other. Therefore, there is no current in the balancing-network loop. Signal power is sent to the output of the E–W amplifier and to the input of the W–E amplifier. The signal reaching the E–W amplifier is dissipated; the signal reaching the W–E amplifier has its power level raised for further transmission.

c. Connection of Four-Wire Terminating Sets in Line. The four-wire terminating set (fig. 72) can be used *at the terminals of a four-wire system,* for it has the necessary four-line connections on one side and two switchboard connections on the other. Figure 73 shows how two four-wire ter-

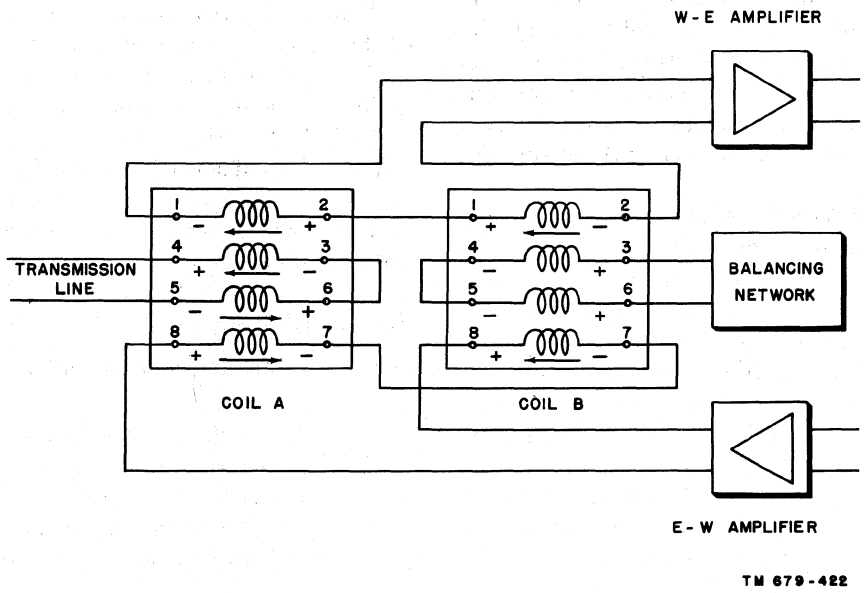


Figure 72. Four-wire terminating set in transmission from line to W–E amplifier.

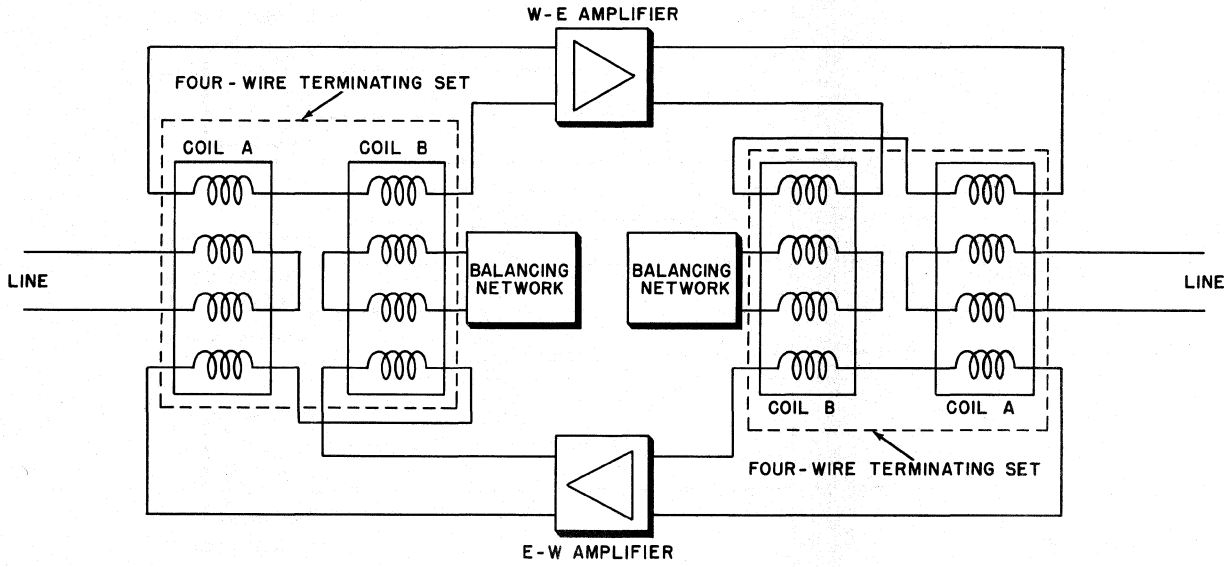


Figure 73. Arrangement of four-wire terminating sets in two-wire repeater.

minating sets are connected to the amplifiers in a repeater of a *two-wire repeater system.*

48. Practical Coil Construction

a. Lack of Perfect Balance. In the hybrid arrangements (pars. 40 through 47), the coils are assumed to be ideal. Ideal hybrid coils, however, do not exist. Manufacture is difficult, and coils cannot be made identical. At best, a high degree of balance is attained, and this suffices. Some signal reaches the input of one amplifier from the output of the other, but the usefulness of the hybrid arrangement is not destroyed as long as this signal is very small. The presence of this undesired signal limits the amount of amplification that can be provided by a repeater without singing, however.

b. Symmetrical Hybrid Arrangements. The three types of hybrid arrangements (pars. 41, 42, and 43) are symmetrical. This means that the signal power in *db* lost is the same in both directions of transmission: the loss in *db* of signal power betweeh the line and the amplifier input is equal to that between the other amplifier output and the line. Only ideal hybrid coils would meet these conditions exactly, and, consequently, with practical ones the transmission loss is not the same in both directions. In well constructed hybrid coils, the difference is very small, however—approximately .1 *db*.

49. Transmission Loss in Hybrid Arrangements

As applied to transformer-type hybrids, *transmission loss* is the loss in signal power through hybrid action plus coil-core and winding-resistance losses. In other words, it is the change in the level of signal-power for line-to-amplifier-input transmission, or for amplifier-output-to-line transmission. In the three hybrids discussed, the transmission loss, for all practical purposes, is the same. In either direction of transmission, the initial signal is divided into two almost equal parts. One half of the signal is completely absorbed, and the other half is used for further transmission. This hybrid action causes a 3-*db* loss. In addition, the core and winding of the coil absorb a small amount of power. This absorbed power is from .25 *db* to 1 *db*. Hence, a close approximation of the transmission loss in either direction is about 3.25 *db*. The amplifier readily makes up this loss.

50. Transhybrid Loss

a. Loss from Mismatch of Balancing Network and Line.

(1) The ability of the hybrid arrangement to keep power from the output of one amplifier out of the input of the other depends on the similarity of the line and balancing-network impedances. In the

ideal case, there is no power at the amplifier input; there is an infinite loss between the output of one amplifier and the input of the other. In practice, this cannot be accomplished, because it would require line and balancing-network impedances to be exactly the same at all frequencies. Some power, therefore, always reaches the amplifier input. This loss—of signal power between the output of one amplifier and the input of the other amplifier—is called the *transhybrid loss*. Any power that does *not* reach the amplifier input is considered *lost power*.

(2) The signal power from the amplifier output divides about equally between the line and balancing network, giving a loss of 3.25 *db*. Assuming that the line impedance differs from the network impedance, a portion of the power entering the line is reflected back toward the hybrid coils, as described in chapter 2. The power not reflected is known as the *R.L.* (return loss). The *current reflection coefficient* is the ratio of reflected current to incident current at any given point in a transmission line. The reflection coefficient is: $(Z_B-Z_C)/(Z_B+Z_C)$.

(3) Figure 74 illustrates the application of this formula. Z_B is the internal impedance of the generator and Z_C is the transmission-line impedance. The relationship between the incident power P_i and the reflected power P_r is

$$P_r=\left|\frac{Z_B-Z_C}{Z_B+Z_C}\right|^2 P_i,$$

The value of the return loss is thus a measure of the similarity between the line and network impedances; the greater the similarity, the greater the return loss. Note that for exact balance Z_N equals Z_L, that the denominator of the fraction $\left|\frac{Z_N+Z_L}{Z_N-Z_L}\right|$ is zero, and which reads: The reflected power equals the product of the absolute value of the magnitude of the reflection coefficient squared and the incident power. The | | sign indicates the numerical or absolute value of the quantity within it. In this hybrid arrangement we can consider the balancing network as the generator impedance. Since Z_N and Z_L are the balancing-network and line impedances, respectively,

$$P_r=\left|\frac{Z_N-Z_L}{Z_N+Z_L}\right|^2 P_i \text{ or } \frac{P_i}{P_r}=\left|\frac{Z_N+Z_L}{Z_N-Z_L}\right|^2.$$

The return loss in decibels is

$$R.L.=10 \log \frac{P_i}{P_r}=10 \log \left|\frac{Z_N+Z_L}{Z_N-Z_L}\right|^2,$$

or

$$R.L.=20 \log \left|\frac{Z_N+Z_L}{Z_N-Z_L}\right|.$$

that the value of the fraction is infinite. Thus, the return loss is infinite. The power reflected back to the hybrid coil undergoes hybrid action again; that is, half of this power reaches the amplifier input and there is another 3.25-*db* loss. Since the transmission loss is 3.25 *db*, the total loss in this path is the sum 3.25 *db* plus *R. L.* plus 3.25 *db* or *R. L.* plus 2 times the transmission loss. Return loss is readily calculated. For example, assume a line impedance of 440 ohms and a balancing network impedance of 360 ohms.

$$\text{Then } R.L.=20 \log \left|\frac{360+440}{360-440}\right|=20 \log 10$$

$$R.\ L.=20\ db.$$

Note. If the answer indicates a negative log, as in this case 20 log—10, merely delete the minus sign. This is necessary since only the *absolute* value of Z_N-Z_L is desired. Adding twice the transmission loss to the return loss, the transhybrid loss is

$$20\ db+2(3.25\ db)$$

or

$$26.5\ db.$$

Since both Z_N and Z_L may vary with frequency, a return-loss measurement or computation must be made in terms of a single frequency. The gains and losses in the measuring circuit are expressed for the particular frequency used. Measurements or computations must be made for a number of frequencies in the voice range, to determine at what point in the range the balance conditions are poorest.

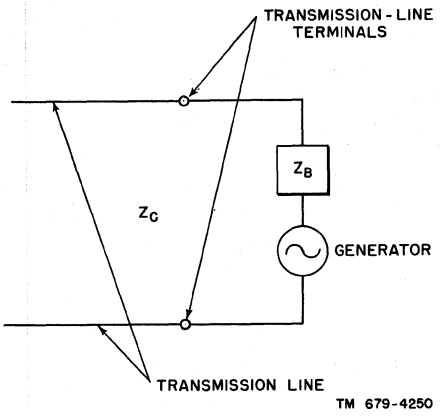


Figure 74. Power reflection.

b. Loss from Mismatch Along Transmission Line. A mismatch of impedances causes an undesirable decrease in the transhybrid loss (*a* above). This mismatch occurred at the terminals of the hybrid set. When the impedance mismatch is at some point along the line, part of the signal power is reflected back toward the hybrid set, and there is an additional line loss that has to be added, to ascertain the total transhybrid loss. For example, let l be the distance between the hybrid set and the point of mismatch in miles, and a be the loss in *db* per mile of the line over which the reflected signal travels; let Z_1 be the characteristic impedance of the line from which the signal approaches the mismatch, and Z_2 be that of the line beyond the mismatch. The additional loss being 2 la *db*, the total return loss in *db* is

$$R.L. = 20 \log \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| + 2\ la.$$

Thus, the total transhybrid loss is

$$20 \log \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| + 2\,la + 2 \text{ times the transmission loss}$$

51. Resistance-Hybrid Arrangement

A *resistance-hybrid arrangement* consists of four equal center-tapped resistors (fig. 75). It does not give as high a degree of balance as the hybrid coil or repeating-coil hybrid, and has a greater transmission loss.

a. Theory of Operation of Resistance Hybrid. The d-c Wheatstone bridge provides the basis principle of the resistance hybrid. Turn to figure 58, which shows the Wheatstone bridge arrangement, and imagine R_3 replaced with a transmission line and R_4 with a balancing network. An unbalanced line condition results; therefore, if the bridge principle is to be used, it must be modified. Let the modification take the form of dividing each of the 4 resistances into 2 equal parts. Refer to figure 75, where this has been done. Momentarily, view this figure as though the balancing network and transmission line were not shown. For a signal applied across D–E, there will be no difference of potential across D–F, which indicates that the placement of the amplifiers is correct (the output of one does not feed into the input of the other).

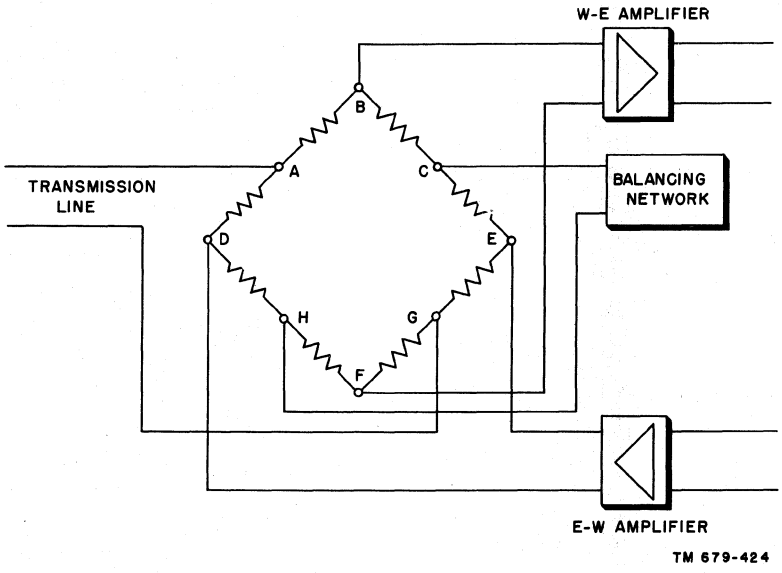


Figure 75. Resistance-hybrid arrangement.

Now, the transmission line must be placed across points of different potential in such a manner that the line is terminated in a symmetrical network. Two points that meet this condition are A and G. Place the line across A–G. This unbalances the bridge, but placement across C–H of a balancing network equal to the line impedance restores the balance of the bridge, making the hybrid operable. This hybrid set is a closed circuit to direct current, which restricts its use to lines which do not carry direct current. Figure 76 is the conventional way of showing the circuit of the resistance hybrid. Note that corresponding points in this figure and in figure 75 have the same identification.

b. Losses of Resistance Hybrid. The resistance hybrid has a transmission loss of approximately 9.5 *db*. This is greater than the transmission loss of the transformer-type hybrid, because the resistors dissipate more power than the coils. There is also a transhybrid loss, but this is less than the loss in the transformer type because of the poorer balance obtained.

52. Types of Balancing Networks

The impedance of a line or cable connected to a hybrid set must be balanced by the impedance of a network also connected to the hybrid set. The extent to which the repeater may improve transmission depends directly upon the degree to which the network balances the line. Since the line impedance varies with frequency, the balance must cover the frequency range of the repeater. Various types of *balancing networks*, which come in a number of standard arrangements, are used in practice, choice of any one depending on the line construction (whether loaded or unloaded cable, or open wire line is used) and the type of line termination used at the terminals.

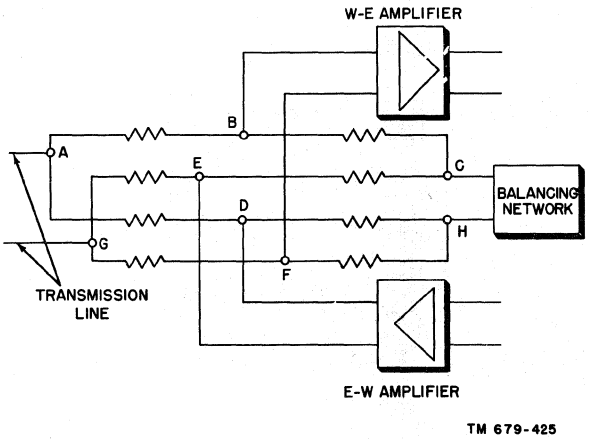


Figure 76. Resistance-hybrid arrangement, conventional diagram

a. Compromise Network. Since the impedance looking into a switchboard varies, depending on the circuit to which it is connected, the switchboard and balancing network, functioning alone, cannot have a high degree of balance with all impedances. Improvement is effected, however, by inclusion of a *compromise network*—a network that brings toward an average the impedances seen by the switchboard. Figure 77 shows a compromise network. It consists of a capacitor, of approximately 2 microfarads, in series with a resistor, usually of 600 ohms.

b. Line-Balancing Networks.

(1) *Basic network.* A high degree of balance is attainable in long lines between repeaters because of their fixed length superior construction, and the fact that they are usually terminated in their characteristic impedance. To balance their impedance, a *basic network* is employed. Two types of basic networks are in use: precision networks and adjustable networks.

(2) *Precision network.* Line construction has become highly standarized, and standard balancing networks, already assembled and wired, are available, ready for use with all of the standard types of construction, both open-wire and cable. Such a standard balancing network, employed to balance a standard type of line construction, is called a *precision network.*

(3) *Adjustable network.*

(*a*) An *adjustable network* is used in a line that cannot be balanced with a precision network. Such a line may be one of nonstandard construction; one of standard construction, but short; or one having large irregularities of impedance, as might occur in an open-wire line containing a section of cable, or a line that is not terminated at

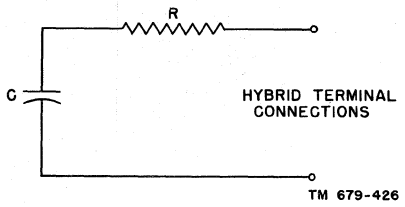


Figure 77. Compromise network.

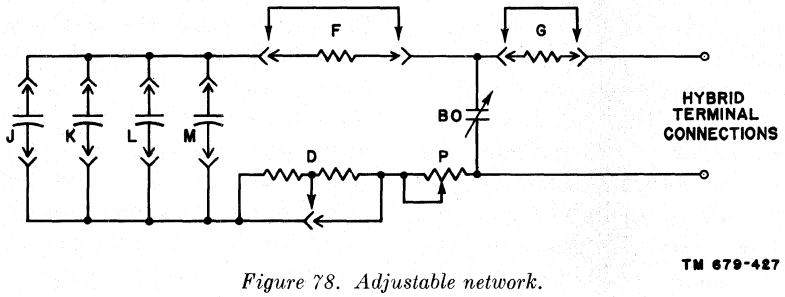


Figure 78. Adjustable network.

either end in its characteristic impedance. The adjustable network is supplied mounted and partially wired; the wiring is completed after thorough testing and adjustment at the time of installation. Figure 78 shows an adjustable network. Resistors F, G, and D, and capacitors J, K, L, and M are combined as requred, and fine adjustments are made with potentiom eter P and variable capacitor BO.

(*b*) Figure 79 is the front view of a typical balancing-network panel. The unit contains two adjustable balancing networks, labeled WEST NET and EAST NET. Each is used in conjunction with a separate hybrid arrangement. Figure 80, a rear view, shows how the many wires and terminals are kept in orderly arrangement by means of terminal plates. The unit is designed for standard 19-inch rack mounting.

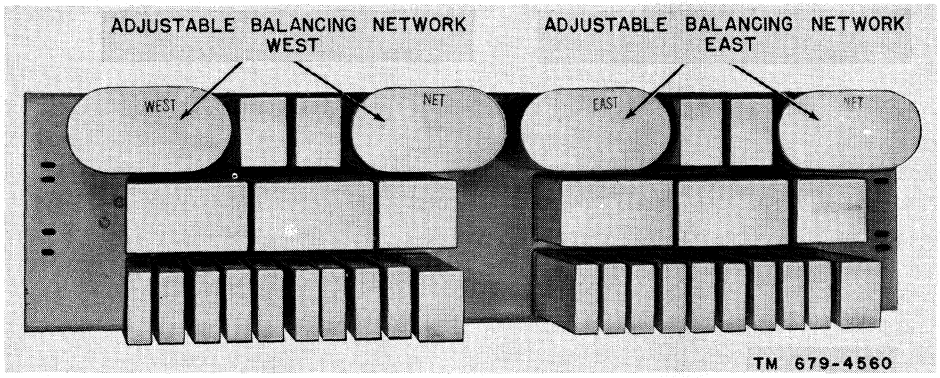


Figure 79. Balancing-network panel, front view.

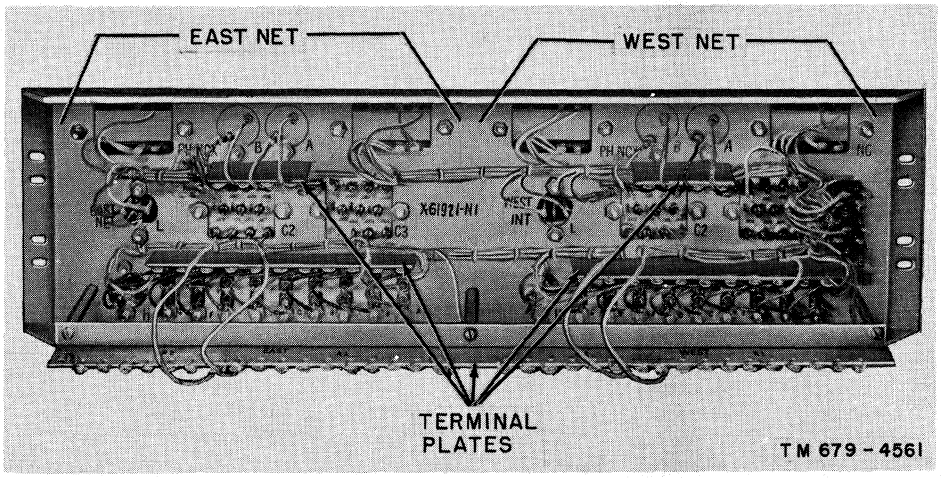


Figure 80. Balancing-network panel, rear view.

c. Building-Out Network. A *building-out network* is used when a repeater station must be connected to the termination of a distant open-wire line by cable—a situation occurring in large cities and other locations where the erection of wire lines is not feasible or not permitted. The building-out network may consist of a single capacitor or an arrangement of a capacitor with one or more resistors.

d. Line-Equipment Balancing Network. The basic network which balances the line does not balance the line equipment. Line equipment requires a *line-equipment balancing network.* For example, composite sets and line filters require composite balancing sets and line-filter balancing equipment. Composite sets and line filters are standardized, and standard line-equipment balancing networks are available for use with them. Line equipment, when used, is placed between the line proper and a hybrid arrangement.

e. Complete Network. A single arrangement may be made combining all the various types of balancing networks except the compromise network. This combination is called a *complete network.* In a complete network the individual networks are arranged in a mirror image of the line and line equipment (fig. 81). As seen from the hybrid arrangement, the components of the complete network, to the right, are arranged in the same order as the composite-set and transmission-line components, to the left.

53. Use of Hybrid Sets

The hybrid sets can be classified by their inherent degree of *balance.* Because of their construction, the hybrid coil and the repeating-coil hybrid are able to give a higher degree of balance than the four-wire terminating set and the resistance hybrid. In a given system, the degree of balance that is desired dictates the type of hybrid set to be used. In a four-wire system that is converted to two-wire operation only at the terminals, hybrid sets that give a lesser degree of balance can be used; but in a two-wire system which employs several repeaters, hybrid sets giving a larger transhybrid loss are necessary for each four-wire to two-wire conversion. This is because there is a greater number of feedback loops in such a system, and the total power fed back toward the sending end from each hybrid set must be as small as possible. Otherwise, the complete system may sing, making communication difficult or impossible.

54. Amplifiers

Repeaters may or may not make use of hybrid arrangements, but the four common types of repeaters do have one feature in common—they employ one or more amplifiers to increase the signal power. An amplifier may have one or more stages of amplification; that is, it may have one or more vacuum tubes connected in cascade (series) with a coupling network or networks. When a single stage of amplification does not give sufficient gain, an additional stage, or stages, is used to obtain it. Typical values of circuit elements of repeater amplifiers and typical operating performance are indicated by the two amplifier circuits actually used in repeaters described in paragraphs 55 and 56. The first circuit is a single-stage amplifier; the second is a two-stage amplifier.

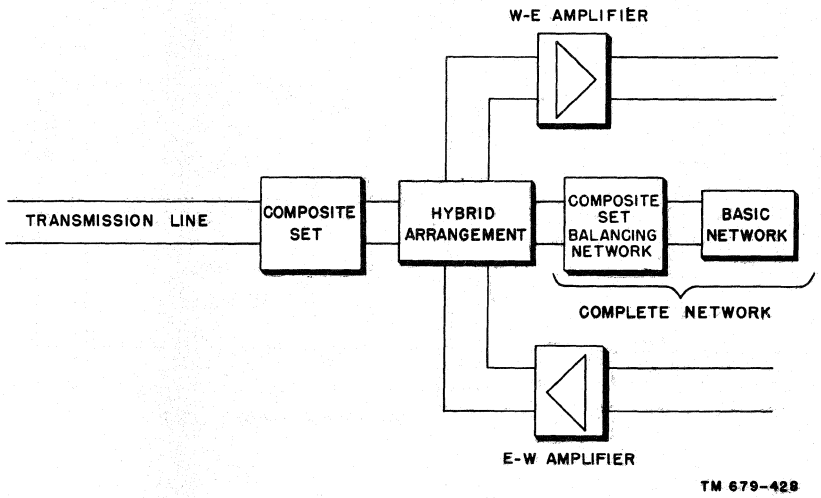


Figure 81. Complete network.

282463 O—54——5

Both are applications of basic principles of vacuum-tube amplifiers discussed in TM 11–662 and TM 11–670.

55. Single-Stage Amplifier

a. Description of Single-Stage Amplifier. A *single-stage amplifier* is shown in figure 82. It has three major components—the input transformer IN, the pentode tube and associated networks, and the output transformer OUT. The input transformer has two 300-ohm resistors in series across its fixed primary winding 8–9, and a multiple-tap secondary winding 1–7. The signal voltage across the secondary winding is distributed across the grid-cathode impedance of the tube, the series potentiometer GC, and the series resistors C and B. The output of the tube appears across primary winding 7–8 of the output transformer, where it simultaneously induces signal into the other three windings.

b. Operation of Single-Stage Amplifier.

(1) The resistances bridged across primary winding 8–9 of the input transformer fix the input impedance of the amplifier at approximately 600 ohms (fig. 82). This is done to match a 600-ohm transmission line, or 600-ohm network. The secondary winding of the input transformer is connected to the control-grid cap of the 310A pentode tube at the first of the six taps on the winding. Changing the tap on the secondary winding causes less signal voltage to be applied to the tube; this is one way to adjust the gain of the amplifier. The taps are arranged so that adjustments for gain can be made in steps of 4 *db*. This is a coarse gain adjustment, however. For fine-gain adjustment, potentiometer GC in the tube cathode circuit is varied.

(2) Cathode current having a d-c component flows through resistors B and C and potentiometer GC, causing a potential difference across them (fig. 82). This potential difference is the negative bias voltage for the control grid. Changing the setting of potentiometer GC so that more or less resistance is included in the circuit changes the bias voltage, and so changes the amplifier gain over a small range of 5 *db*. Maximum gain through this small range is obtained when the potentiometer is so set that there is no resistance of the potentiometer in the

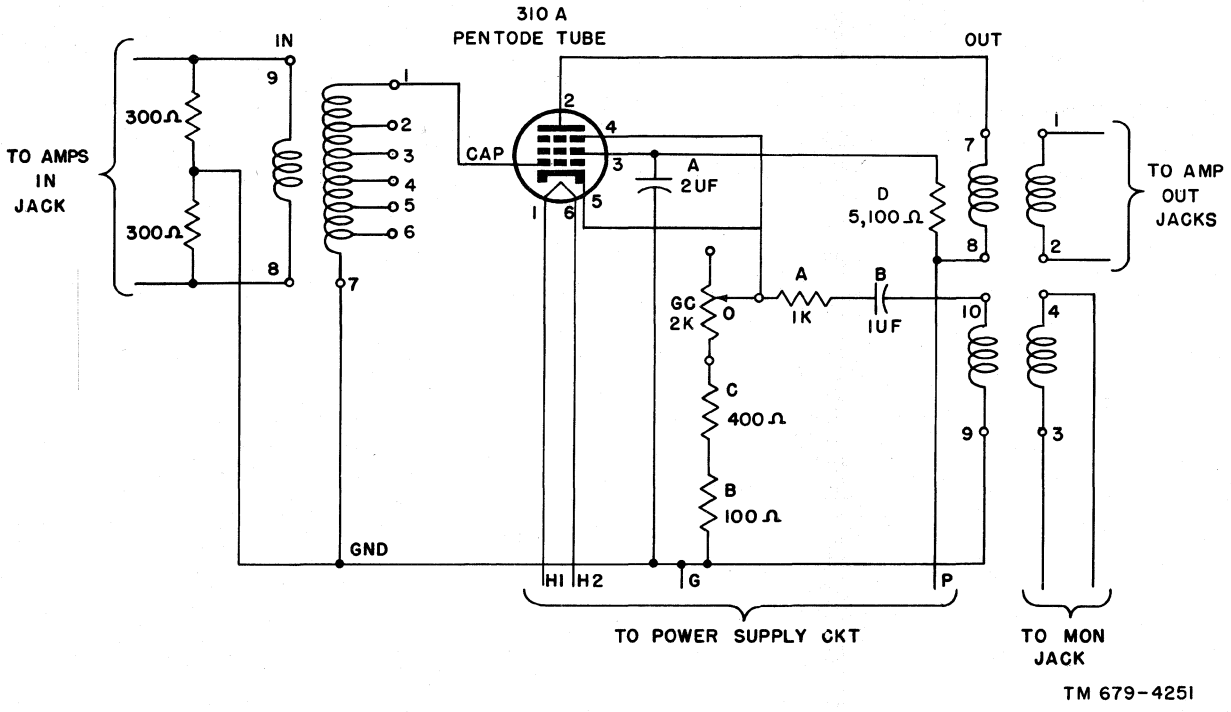


Figure 82. Single-stage amplifier.

circuit. A maximum amplifier gain of 33 *db* is obtained with the control grid connected to tap 1 of the input transformer and GC is set for maximum gain. The output of the tube appears across primary winding 7–8 of the output transformer. Signal then is induced into windings 1–2, 3–4, and 9–10. The signal from winding 1–2 is the output signal of the amplifier, and is sent into the outgoing transmission line.

(3) Winding 3–4 is called the monitoring winding. The output from it is 11 *db* below the output of the amplifier at winding 1–2. This signal goes to a telephone receiver for the use of the repeater operator.

(4) The amplifier also employs a negative-feedback circuit (fig. 82). The negative-feedback circuit consists of negative-feedback winding 9–10, capacitor B, and resistor A. The negative feedback voltage is applied across potentiometer GC, and resistors C and B. This feedback circuit makes the gain relatively independent of the usual fluctuations in power-supply voltage, and also makes the output impedance constant at approximately 600 ohms over most of the voice-frequency range.

(5) Capacitor A is a screen bypass and is used to keep the signal from feeding back into the power supply. Resistor D is a screen-dropping resistor.

(6) The gain of this typical amplifier is flat, within about .5 *db* for frequencies between 300 and 3,000 cycles.

56. Two-Stage Amplifier

a. Description of Two-Stage Amplifier. A *two-stage amplifier* is shown in figure 83. It differs from a single-stage amplifier in that it has two vacuum tubes and two input transformers, and has a different negative-feedback circuit. The two-stage amplifier has five major components—the input transformer IN A, the pentode tube V1 with associated networks, the input transformer IN B, the pentode tube V2, and the output transformer OUT. The input transformer has resistors A and B across its primary winding 1–2, and has a secondary winding 7–8. This secondary winding feeds the incoming signal to the control grid of the pentode tube V1. The cathode is connected to the other end of the secondary winding by series

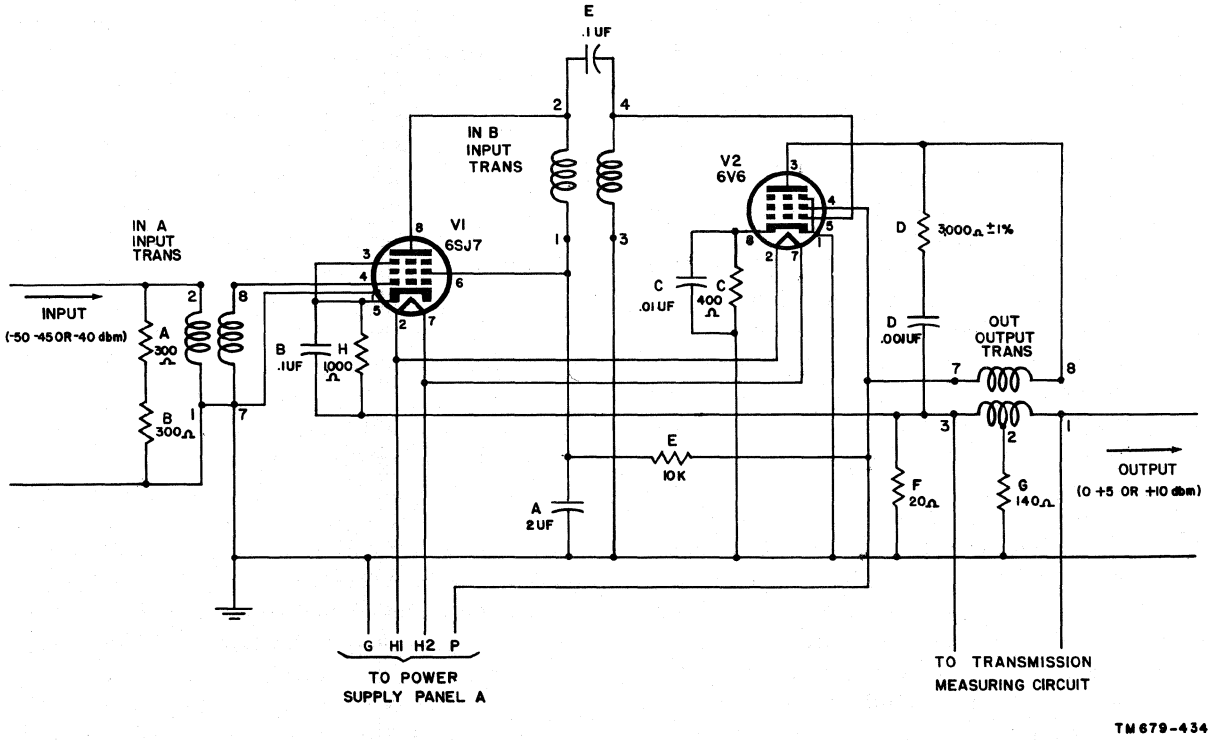


Figure 83. Two-stage amplifier.

resistors H and F. The output of the tube is fed to the primary winding 1–2 of input transformer IN B. Secondray winding 3–4 of this transformer feeds the signal to the control grid of the beam power tube V2. The cathode is connected to the other end of the winding by cathode resistor C. The tube output is fed to primary winding 7–8 of output transformer OUT. Secondary winding 1–3 of the output transformer is tapped at terminal 2.

b. Operation of Two-Stage Amplifier.

(1) In figure 83, the same plate supply P, is used for both stages. To insure that the common plate supply does not cause poor amplifier operation, a plate-decoupling network is used. The plate-decoupling network consists of resistor E and capacitor A. Resistors A and B, across primary winding 1–2 of input transformer IN A, fix the input impedance of the amplifier at approximately 600 ohms. This is done to match a 600-ohm network. Should a 600-ohm transmission line be connected directly across resistors A and B, the ground connection would be removed from terminal 1 of IN A, and placed at the junction of resistors A and B, so that the transmission line would be balanced to ground; otherwise it would not be balanced to ground. Secondary winding 7–8 of IN A is connected across the control grid of V1 and ground.

(2) To understand how signal is impressed on the control grid of tube V1 from secondary winding 7–8 (fig. 83), it is necessary to trace the circuit from control grid to cathode. Starting at the control grid (pin 4 of tube V1) the circuit is through winding 8–7 to ground, from ground through resistor F to the self-biasing combination of capacitor B and resistor H, and through this parallel combination to the cathode (pin 5 of tube V1). Thus, signal is supplied to tube V1, and the amplified signal appears across primary winding 1–2 of input transformer IN B. The induced voltage across the secondary winding 3–4 of IN B is applied to the control grid of tube V2 (pin 5). The circuit from the control grid to the cathode of tube V2 is through secondary winding 4–3 to ground, and, from ground, through the self-biasing parallel combination of capacitor C and resistor C, to the cathode. Capacitor E and the transformer coils form a combined inductive-capacitive coupling network. The effect of this coupling network is to make the gain practically constant over the entire frequency range being used. The mutual coupling varies inversely with the frequency. This compensates for the increased gain of a transformer coupled amplifier at the higher frequencies. Once more the signal is amplified and appears across the primary winding 7–8 of output transformer OUT. Secondary winding 1–3 is tapped at terminal 2 and resistor G is connected from there to ground. The transmission measuring circuit leads to terminals 1 and 3 and then are balanced to ground. The signal across the 1–3 secondary winding is applied to the transmission line through resistor F. A small portion of the amplified signal voltage thus appears across resistor F.

(3) Since resistor F is in the control-grid-to cathode circuit of V1, negative feedback to tube V1 occurs. The gain of the amplifier, therefore, is independent of the usual fluctuations in power-supply voltage, and the output impedance tends to be constant.

(4) Resistor D and capacitor D, in series with resistor F, are essentially in parallel with the 7–8 primary winding since terminal 7 is connected to ground by a large capacitor in the power supply. The reactance of capacitor D is *smaller* at the higher frequencies than at the lower frequencies. This compensates for the higher reactance of the 7–8 primary winding at the higher frequencies. The effect of resistor D and capacitor D is, then, to equalize the gain of tube V2 for the entire band.

(5) The gain of the typical amplifier shown is about 50 *db* over the frequency range of 200 to 12,000 cycles. Note that there are no variable circuit elements in this amplifier; hence, there is no direct means of varying the amplifier gain. It is effected by placing a variable attenuator before the amplifier in the repeater; this causes the input signal level to the to the amplifier to be varied and conse-

quently the amplifier output is also varied.

c. Amplifiers in Four Repeaters. For the four common types of repeaters, (fig. 84), the 21-, 22-, 44-, and terminal-types, see paragraphs 57, 58, 61, and 62. All but the 44-type repeater use hybrid arrangements. All have one or more amplifiers, as shown in block form. It should be noted that the repeaters may have additional networks, such as filters and equalizers, which are not shown.

57. 21-Type Repeater

a. Description. The 21-type repeater (A, fig. 84) employs a single amplifier and a single hybrid coil, and has a circuit arrangement which requires no balancing network. The balancing network is dispensed with by an arrangement that causes the line on one side of the repeater to be balanced by the line on the other side. Approximately one-half of the signal power entering the repeater from either line reaches the amplifier input. The other half is dissipated in the plate circuit. The amplified signal voltage appears across winding A–B and induces signal voltage into windings C–D and E–F. The instantaneous voltage polarities across the windings are as indicated. The amplified signal thus is divided equally between the two transmission lines, one-half of the power traveling toward the listener, the other half returning to the speaker. The speaker hears his voice as an echo. This feature limits the quality of the communications that can be obtained.

b. Application. The stability (freedom from singing) of the 21-type repeater depends on the

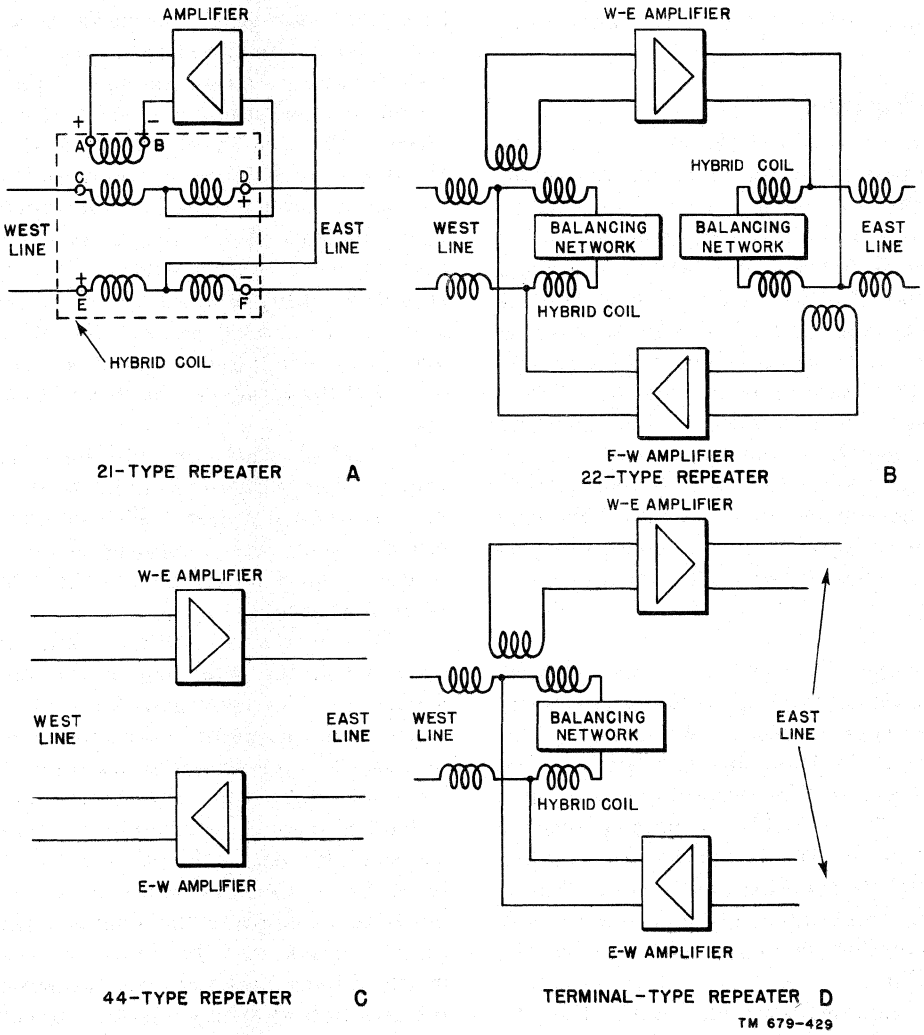


Figure 84. Four common types of repeaters.

assumed equality of the transmission-line impedances on the two sides of the repeater. Because of this, the 21-type repeater is not suitable for use in circuits made up of more than one kind of line. The best location for this repeater is at the midpoint of a circuit. With similar wires on both sides of it, changes of impedance with weather tend to be the same, thus helping to maintain stability. This is an advantage where repeaters are required on nonstabilized lines (lines for which capacitance, leakage, and loss increase by large amounts from dry to wet weather). It is possible to use a 21-type repeater on a loaded circuit, but the usable gain is subject to wide variations, and in some cases may be very small. The 21-type repeater, having only one amplifier, is readily operated by means of dry cells. This has the advantages that the unit may be compact, and that it does not require continuous attention and adjustment after installation.

58. 22-Type Repeater

a. Advantage. Use of the 22-type repeater (B, fig. 84) is not limited by its circuit arrangement, as is true of the 21-type repeater. The 22-type repeater employs a hybrid arrangement and a balancing network for each line, and it is because of this that lines of different impedances may be connected to the repeater without causing singing. The figure shows a hybrid coil but, in general, any one of the 4 hybrid arrangements might be used. Singing or near-singing conditions—both of them harmful to transmission—can be present in a 22-type repeater, however, when there is poor balance between a transmission line and its balancing network. Experimental tests, therefore, are performed after installation, to determine the degree of balance.

b. Operation.

(1) Knowledge of hybrid action permits ready understanding of the operation of the 22-type repeater. An incoming signal from the West line undergoes hybrid action at the West hybrid coil, as shown in B. The part of the signal reaching the input of the W–E amplifier is amplified and sent to the East hybrid coil. There, hybrid action takes place once more, and the amplified signal is sent out on the East line.

(2) For a numerical example, assume that the level of the incoming signal at the input of the West hybrid coil is −30 *dbm*. This level is reduced to −33.5 *dbm* upon passing through the hybrid coil to the W–E amplifier input. An amplifier gain of 43 db gives a +9.5-*dbm* signal level at the output of the W–E amplifier. Signal power at this level enters the East hybrid coil. There, a 3.5-*db* transmission loss reduces the level to +6 *dbm* at the East line terminals for further transmission.

(3) East-to-West operation is similar, the signal path passing through the E–W amplifier.

59. Singing of 22-Type Repeater

a. Limitation Imposed by Singing of 22-Type Repeater. The operation of a 22-type repeater has a limitation. In practice, the hybrid construction is not perfect, and networks never balance the transmission line perfectly at all frequencies. Thus, the transhybrid loss (par. 50) is not infinite, and a certain amount of power from the output of one amplifier is fed to the input of the other. The extent of the *feedback* depends on the degree of balance; the greater the unbalance, the greater the feedback. The amount of feedback, in turn, limits the amount of gain which may be used at a repeater. The limits may be stated in terms of a simple rule: *When the total gain in the circuit between the W–E and E–W amplifiers of a repeater equals or exceeds the total loss of that circuit, enough feedback exists to cause singing.*

b. Singing Path of 22-Type Repeater. The broken lines in figure 85 indicate the *singing path* of a 22-type repeater. The singing path is the circuit among the amplifiers of one or more repeaters, through which flow the circulating currents that give rise to *singing*.

c. No-Singing Condition. Singing does not occur when the total loss in the singing path is *greater* than the total gain (fig. 85). A signal introduced at any point in the singing path decreases in intensity each time it goes around, finally reaching zero. The time this takes depends on the difference between the total loss and total gain in the singing path. The greater the difference, the fewer the number of times the circulating signal needs to travel over the singing path to be completely dissipated. In figure 85, the only losses in the singing paths are the transhybrid losses of the two hybrid arrangements. Recall that transhybrid loss is the loss of signal between the output of one amplifier and the input of the other ampli-

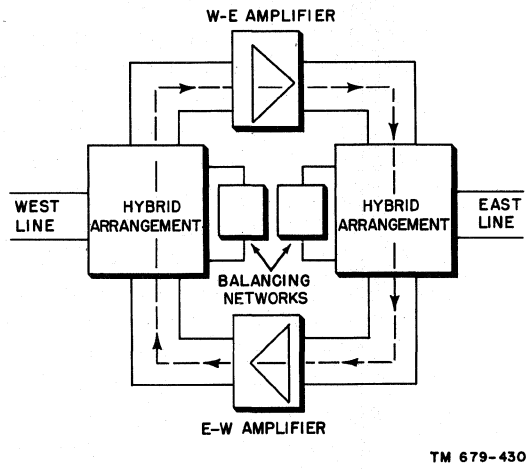


Figure 85. Singing path of 22-type repeater.

fier in the same repeater. The greater this loss, the more effective is the repeater. Assume that the transhybrid losses for the West and East hybrid arrangements are 32 *db* and 35 *db*, respectively, and that the gains of the E–W and W–E amplifiers is 31 *db* and 29 *db*, respectively. In the singing path, the total loss is 67 *db*, and the total gain is 60 *db*. Since the total loss is greater than the total gain (7 *db* greater), singing does not occur.

d. Singing Condition. Singing occurs when the total loss in the singing path is *less* than or equal to the total gain (fig. 85). A signal introduced at any point in the singing path *increases* in intensity each time it goes around. The level of the circulating signal, therefore, increases to a maximum determined by the power capabilities of the amplifiers, and singing occurs. Assume that the transhybrid losses for the West and East hybrid arrangements are 32 *db* and 35 *db* respectively, and that the gains of the E–W and W–E amplifiers are 36 *db* and 33 *db* respectively. In the singing path, the total loss is 67 *db*, and the total gain is 69 *db*. Since the total loss is *not* greater than the total gain (2 *db* less), singing occurs.

e. Near-Singing Condition. If the total loss is only slightly greater than the total gain around the singing path, a near-singing condition called *barreling*, harmful to transmission, is produced. Barreling exists when words reverberate and sound hollow—as if they were spoken into an empty barrel. It results from the passing of a considerable period of time for the decrease to zero of the signals circulating around a repeater.

f. Singing Margin. To prevent barreling, the total gain in the singing path has to be made considerably less than the total loss. The difference between the total loss and the total gain, when the amplifiers are set for normal operation, is called the *singing margin*. Satisfactory operation requires a singing margin of from 8 to 10 *db*, depending on the equipment.

g. Importance of Hybrid Balance in 22-Type Repeater. Since, for good transmission, the total gain in the singing path must be considerably less than the total loss, it follows that for the total gain to be as high as possible, the total loss must be as high as possible. The total loss in the singing path of the circuit (fig. 85) occurs in the two transhybrid losses; therefore the transhybrid loss should be as large as possible. But the transhybrid loss depends on the degree of hybrid balance; the closer the hybrid balance, the greater the transhybrid loss. Therefore, for large signal-power gains in a repeater, and good transmission, the hybrid arrangements must be well balanced.

60. Singing-Point Tests

Because the balance at the hybrid arrangements is so important, efficient operation requires an accurate means for ascertaining it. The degree of balance is ascertained by means of *singing-point tests*, which make use of the practical rule that a repeater sings when the total loss in the singing path is less than or equal to the total gain. The tests are of two kinds: *passive singing-point* and *active singing-point tests*. The only basic difference between them is this: In a passive singing-point test the circuit is arranged so that the measurement depends on the gain of *one* repeater, whereas in an active singing-point test the circuit is arranged so that the measurement depends on the gains of *more than one* repeater.

a. Passive Singing-Point Test.

(1) The passive singing-point test permits calculation of the transhybrid loss of a single hybrid arrangement; and since the transhybrid loss depends on the degree of balance at the hybrid arrangement, this test indicates the degree of balance as well. To determine the transhybrid loss, it is necessary, first, to locate the point at which the repeater begins to sing. Figure 86 shows a typical arrangement for performing a passive singing-point test. The three major components are the repeater under test, the passive repeater, and the line be-

tween the repeaters. In the repeater under test, a calibrated variable attenuator replaces the West hybrid arrangement; this is a device which in use tends to keep the amplifier gains at normal operating values when the singing point is reached. This test, therefore, will determine the degree of balance at the East hybrid arrangement between the balancing network and the line, and will be concerned with a single feedback loop. The passive repeater is the next repeater in the system. It remains connected, in order to terminate the line in the impedance that is present during actual operation without introducing any extraneous signals. To prevent the introduction of extraneous signals by the passive repeater—which would lead to inaccurate measurements—the amplifier gains of the passive repeater are set at zero.

(2) Once this arrangement has been set up, the actual test is simple (fig. 86). First, the tester must find the *singing point*—the point at which the repeater goes from the nonsinging to the singing condition. To do this, he sets the amplifier gains of the repeater under test at their normal operating values. Then he varies the calibrated variable attenuator while listening with the monitor telephone receiver, with which all repeaters are provided. When the singing point is reached, singing, if present, ceases; if singing is not present, it begins. If the repeater is *singing*, and variation of the attenuator over its complete range does not locate a point when the singing *stops*, the tester sets the attenuator for maximum attenuation, and then decreases the amplifier gains until the singing point is reached; if the repeater is *not singing*, and variation of the attenuator over its complete range does not locate a point where singing *begins*, the tester sets the attenuator for minimum attenuation, and then increases the amplifier gains until the singing point is reached. The repeater starts to sing at the voice frequency at which the balance between the balancing network and the line is poorest.

(3) At the point that signing begins, the tester records the amplifier gain values and the calibrated variable-attenuator value (fig. 86). Since the total gain equals the total loss at this point, he can now calculate the transhybrid loss, which is equal to the difference between the total gain of the amplifiers and the recorded calibrated

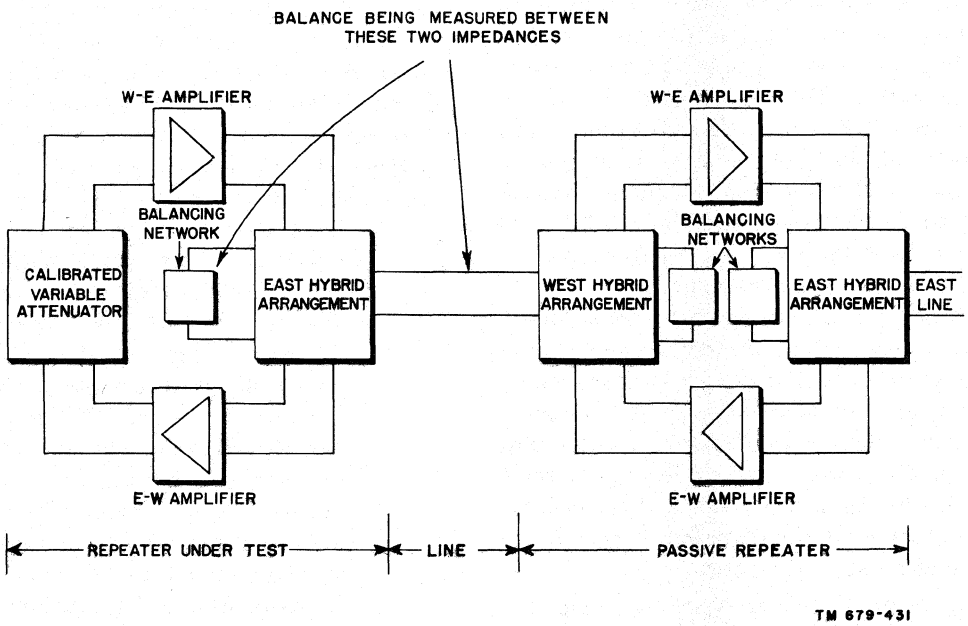


Figure 86. Passive singing-point test.

variable attenuator loss. Performing this test for both of the hybrid arrangements of a repeater gives the total loss in the singing path. Since, for satisfactory operation of a repeater, the singing margin must be at least 8 *db*, the largest total gain in the singing path that can be used is calculated readily. For example, assume that the transhybrid losses are 27 *db* and 29 *db*. The total loss is 27+29 =56 *db*. Thus, a minimum singing margin of 8 *db* permits a maximum total gain of 48 *db* in the singing path.

b. Active Singing-Point Test.

(1) The passive singing-point test gives results that pertain to *one repeater*, not functioning under actual operating conditions. An *active singing-point test* must be performed to determine whether the actual operating conditions of a *repeater system* meet specifications. Almost without exception, active singing-point tests are performed at terminal repeaters—first at one, then at the other. They are concerned with two or more feedback loops. The number of feedback loops in a given system is equal to the number of repeaters in the system.

(2) Figure 87 shows a typical arrangement for performing an active singing-point test. This system consists of a near-end terminal repeater, line section No. 1, an intermediate repeater, line section No. 2, a far-end terminal repeater, and the far-end switchboard. The near-end terminal repeater is the one under test. In it, the West hybrid arrangement is replaced with a calibrated variable attenuator. All the amplifier gains are set at normal operating values. There are three feedback loops, indicated by dashed lines, with arrowheads that show the direction of signal transmission. The feedback loops are: the singing path in the near-end terminal repeater; the path around the near-end and intermediate repeaters, including line section No. 1, not returning in part through the East hybrid arrangement of the near-end terminal repeater; and the path around all the repeaters, including both line sections, not returning in part via shortcuts through the East hybrid arrangements of the near-end and intermediate repeaters. Because there is more than one feedback loop (unlike the situation in the passive test), the near-end terminal repeater will sing at a comparatively lower value of total amplifier gain.

(3) Once the system has been set up, the actual test is simple (fig. 87). First, the tester must find the singing point. To do this, he sets all the amplifier gains at their normal operating values. Next, he sets the calibrated variable attenuator at a value specified for the system; then, listening with the monitor telephone

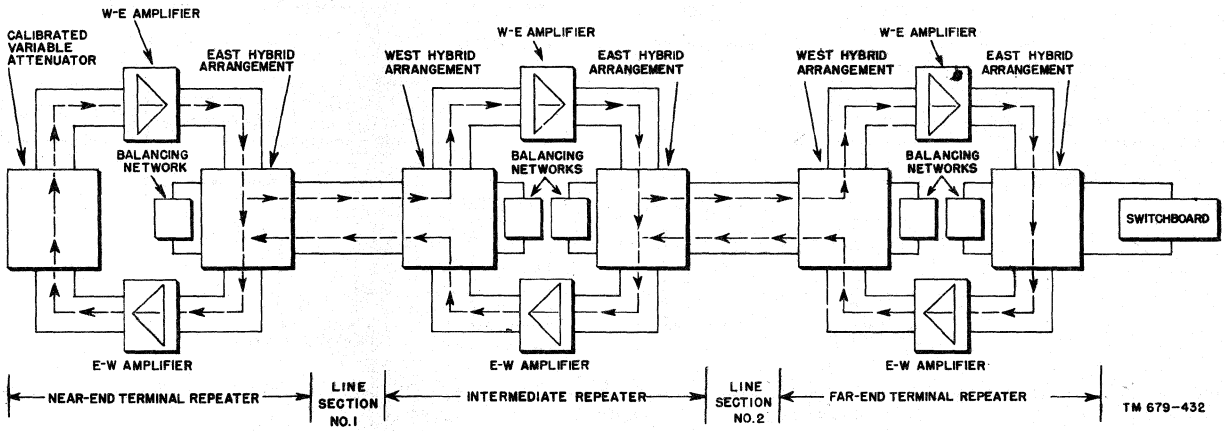


Figure 87. Active singing-point test.

receiver at the near-end terminal repeater, he increases the amplifier gains of the near-end terminal repeater until he detects singing. He records the new amplifier gain values. If they are below the values specified for the system, a passive singing-point test must be made at each repeater, so that the hybrid arrangement or arrangements which are poorly balanced can be located and corrective measures taken.

61. 44-Type Repeater

A 22-type repeater system depends on the degree of balance at the hybrid arrangements for satisfactory operation. The degree of balance limits the amount of amplifier gain that can be used at a repeater. This, in turn, limits the length of the line sections between repeaters and increases the number of repeaters needed between terminals.

a. A repeater system that does not use hybrid arrangements at the repeaters must use a four-wire circuit (C, fig. 84). Each pair of wires is used for a single direction of transmission. The four-wire repeater system employs *44-type repeaters*, except at the terminals, where it must be connected to a two-wire system so that it can be terminated at a switchboard.

b. Figure 88 shows the complete circuit of a four-wire repeater system. It consists of a West terminal, line sections, two 44-type repeaters, and an East terminal. The West and East terminals are identical. Each 44-type repeater consists of two amplifiers which operate in opposite directions. Each of the amplifiers has an input transformer and an output transformer (not shown), to isolate electrically the line sections and keep the line from being unbalanced to ground. Because this system dispenses with hybrid arrangements by using four wires for a single conversation, the amplifier gains are not limited by the characteristics of hybrid arrangements, as are those of the 22-type repeater systems. Because the amplifiers can be set for higher gains, there can be greater distances between repeaters and a smaller number of repeaters between terminals.

c. Since the level of the output signal power is high and the level of incoming signal power of the adjacent pair of wires is very low, crosstalk may result; to prevent this, the pairs of conductors transmitting in opposite directions are kept as widely separated as is practicable.

d. For satisfactory transmission over very long transmission lines, the 44-type repeater system is superior to the 22-type repeater system.

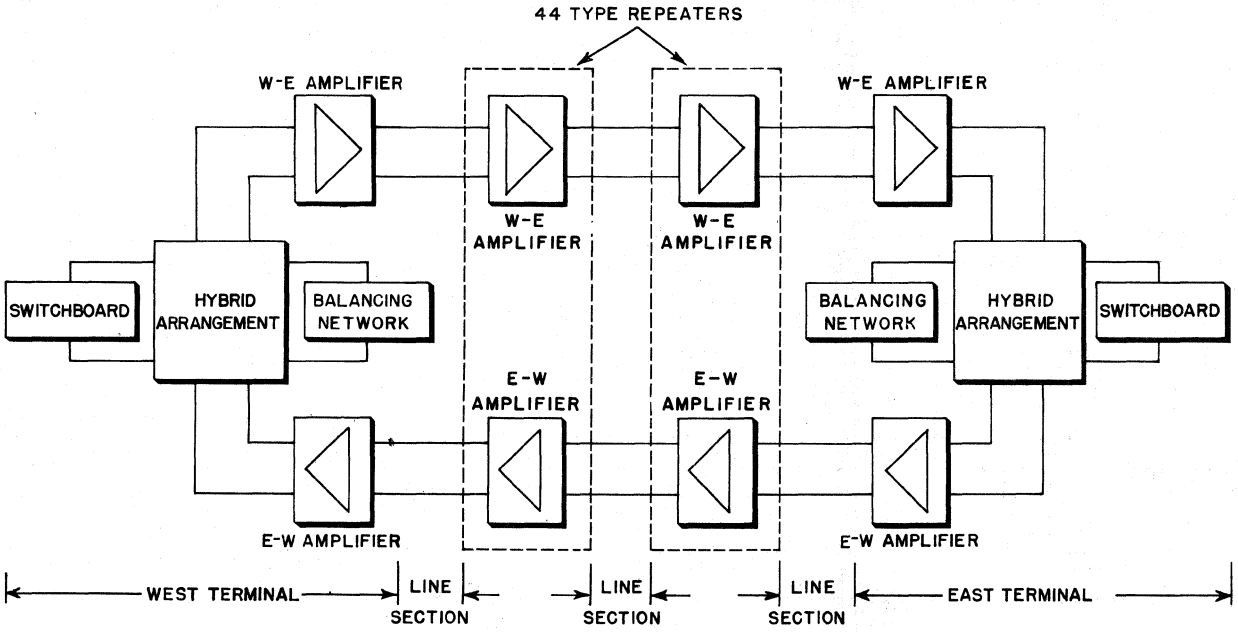


Figure 88. Four-wire repeater system.

62. Terminal-Type Repeater

a. The *terminal-type repeater* is used at the termination of a four-wire repeater system. D, figure 84, shows this to be a 44-type repeater plus a hybrid arrangement. The degree of balance at this hybrid arrangement is determined by the passive singing-point test, as described for the 22-type repeater.

b. When the level of the signal power at a terminal is high, a hybrid arrangement may be used in place of a terminal-type repeater.

63. Spacing of Repeaters

The determination of the proper spacing of repeaters in a repeater system depends on the characteristics of the line or lines to be used, including their noise and crosstalk limitations, and the amplification capabilities of the various repeaters. (For information pertaining to any specific type of system, refer to the manual on the specific equipment.)

a. Line Characteristics. The characteristics of a line include its attenuation under various conditions and its characteristic impedance. Attenuation of open-wire lines is affected by wet or dry weather; in cables, temperature is an important consideration. When more than one type of line is to be employed, it is necessary to know the length and location of each type and the average impedance that is seen looking into the switchboard from the line.

b. Choice of Repeater. The repeater selected must be able to raise the signal level to the desired value without being set for the minimum allowable singing margin under normal operating conditions. This is to permit increase in the repeater output in case of emergency or other need.

c. Length of Line Sections. The spacing of repeaters depends on the line characteristics and the amplification capabilities of the repeaters. When the line and repeater characteristic impedances are matched (as they will be), the desired length of a section of line is equal to the net gain of the repeater (or repeaters) in *db* under normal operating conditions, divided by the line attenuation (of the cable under consideration) in *db* per mile; that is:

(1) Length of line section (in miles)$=\frac{\text{net gain (in } db)}{\text{attenuation (in } db \text{ per mile)}}$. Using this formula, the total attenuation equals the net gain. For an example of how these factors are employed to determine repeater spacing, assume that #19 gage cable is to be used in a certain line, along with identical repeaters which match the line impedance and will provide a net gain of 18 *db* under normal operating conditions. From table II (par. 25*c*), it is learned that the attenuation for #19 gage cable is 1.26 *db* per mile. Substitute these values for the definitions in the equation above.

(2) Length of line section (in miles)$=\frac{18\ db}{1.26\ db \text{ per mile}}=14.3$ miles. Therefore, the length of the line sections should be approximately 14.3 miles. The repeaters would operate properly for line sections of approximately 13 to 15 miles. Much smaller line sections would give good communication, but would be uneconomical in repeater cost, installation, operation, and maintenance. Care must be taken not to make them too long, because the input signal at the repeaters then would be so weak that excessive crosstalk would result. Communication might not only be poor, it might be impossible. The minimum allowable dbm level for signal into a repeater and the maximum allowable dbm level out of a repeater depend on the type of transmission line used in a repeater system.

64. Anti-Hybrid Devices

a. Reversal of Direction of Transmission of Repeater. The performance of a two-wire voice-frequency repeater circuit employing hybrid arrangements depends on the balance of its arrangements. To eliminate the use of hybrid arrangements in this type of circuit, various means, called *anti-hybrid devices*, have been devised. Figure 89 shows a two-wire repeater system, the repeaters of which contain single amplifiers but no hybrid arrangements. Since the amplifier is a one-way device, this system permits one-way transmission only. Two-way transmission would be possible over this system; however, if there were incorporated some means of reversing the direction of transmission of the repeaters as needed during a conversation. In *b* and *c* below, two types of reversing means are devised—the manual (*push-to-talk operation*) and the automatic (*voice operation*). Both types restrict transmission to 1 direction at any one time; that is, while 1 person speaks at 1

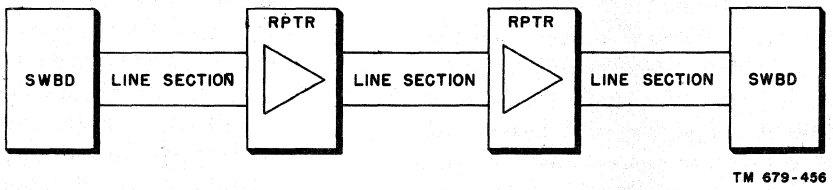


Figure 89. One-way transmission.

end of the circuit, the person at the other end can listen only.

b. Push-to-Talk Operation. In *push-to-talk operation,* repeater reversal is accomplished by means of manual switching at each of the repeater stations of the system. Figure 90 shows the necessary switching arrangement at each repeater. The four-pole double-throw switch is operated by an attendant who monitors the conversation. For transmission from the West line to the East line, the switch is thrown to the left. This connects the West line to the repeater input and the East line to the repeater output. For transmission from the East line to the West line, the switch is thrown to the right. This connects the East line to the repeater input and the West line to the repeater output. The method of push-to-talk operation has the disadvantage of requiring that the losses in each line section be approximately equal. For example, each repeater of figure 89 must compensate for losses in the line section adjacent on *one* side during *one* direction of transmission and on the *other* side during the *other* direction of transmission; and, since the reversal must be rapid for normal conversation, the gain setting of the repeater has to serve for *both* directions of transmission. It also has the disadvantage of requiring the constant attention of personnel at each repeater location, to operate the switches. There is no privacy, and the procedure is slow. This method, therefore, does not give the performance obtained with systems employing hybrid arrangements, and it is used less widely.

c. Voice Operation.

(1) In *voice operation,* repeater reversal is accomplished in each repeater station by means of relays operated by voice frequencies. Figure 91 shows the necessary relay-switching arrangement at each repeater. Relay A is operated for transmission from the West line to the East line, and relay B is operated for transmission from the East line to the West line. Both relays are shown in their nonoperated condition. The operation of one relay open-circuits the other.

(2) For *transmission from the West line to the East line* (fig. 91), voice-frequency currents from the West line operate relay A, which closes contacts 1 and 2 and opens contact 3. This permits the voice signal to enter the repeater input. The repeater

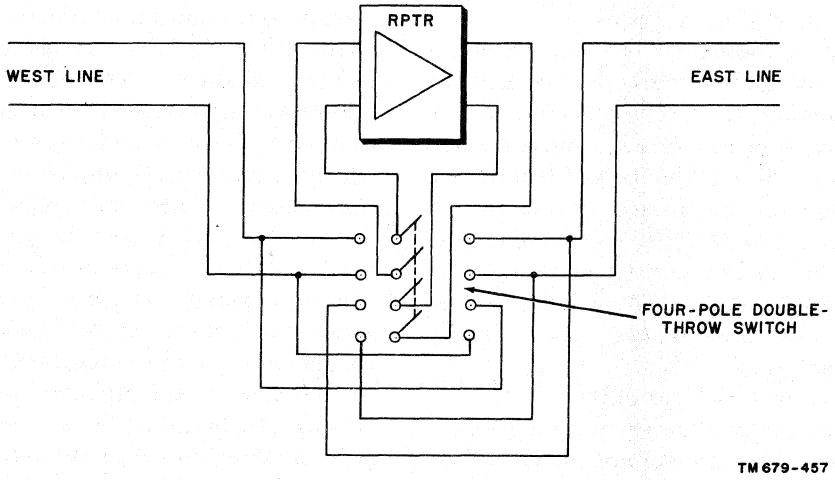


Figure 90. Push-to-talk operation.

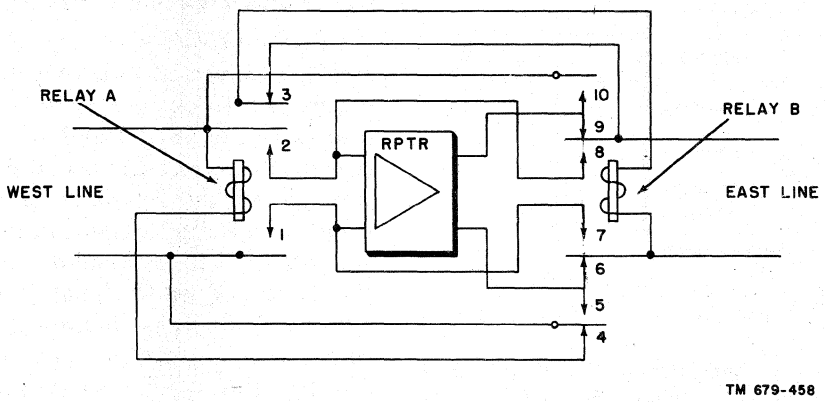


Figure 91. Voice operation.

raises the level of the signal power. The output passes into the East line through contacts 6 and 9 of relay B. Relay B is not operated by voice current because it has been open-circuited by the opening of contact 3 of relay A. Relay A remains operated as long as voice current is present from the West line, after which it returns at once to its nonoperated condition.

(3) For *transmission from the East line to the West line* (fig. 91), voice-frequency currents from the East line operate relay B, which closes contacts 5, 7, 8, and 10, and opens contacts 4, 6, and 9. This permits the voice signal to enter the repeater input. The repeater raises the level of the signal power. The output passes into the West line through contacts 5 and 10 of relay B. Relay A is not operated by the voice current because it has been open-circuited by the opening of contact 4 of relay B. Relay B remains operated as long as voice current is present from the East line, after which it returns at once to its nonoperated condition.

(4) The method of voice operation also requires that the losses in each line section be approximately equal, for the reasons given in *b* above for push-to-talk operation. It, too, has other disadvantages. The person listening cannot interrupt the person speaking. The relays operate quickly, but there is clipping of the first word spoken by each person after a pause. Most important, there may be noise on the line, which may lock out the person on the quieter side of a repeater. On the whole, this method does not give the good and dependable performance obtained with systems employing hybrid arrangements, and it is not used as widely.

65. Terminal-To-Terminal Ringing

Repeaters solve the problem of transmitting voice-frequency signals over very long lines, but they affect the important function of signaling between switchboards. The signaling (ringing) frequency sent out from a switchboard is approximately 20 cycles per second. This is a very low frequency in the voice-frequency band, one not readily amplified by the repeaters. For this reason, very little or none of this signal energy is at the output of a repeater. When there is a small amount, it is not enough to operate the signal-receiving device (drop or buzzer) at the switchboard in the distant terminal. Therefore, the important function of signaling between terminal switchboards is impossible in the repeater system. Signaling can be effected, however, by the provision of additional equipment. The equipment either may bypass the low signaling frequency around the repeater, or convert it to a frequency in the voice frequency band—methods known as 20-cycle ringing and voice-frequency ringing.

66. 20-Cycle Ringing

In *20-cycle ringing* over a voice-frequency repeater system, a 20-cycle ringing signal is transmitted over the entire system. The additional equipment must bypass this signal around every repeater, for both directions of transmission. The equipment takes the form of a network contained in the repeater.

a. Operation. Figure 92 is a block diagram

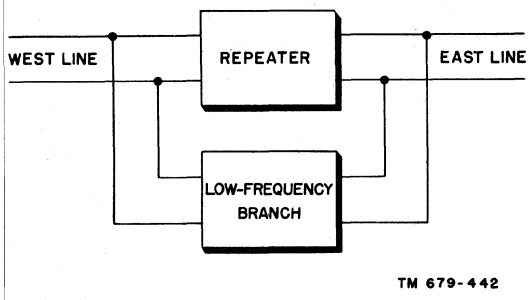


Figure 92. 20-cycle ringing around voice-frequency repeater.

showing how 20-cycle ringing may be accomplished at a two-wire repeater. There are two major components—the repeater and the low-frequency branch. The low-frequency branch is a network that permits low-frequency signals to pass, but blocks all others. A 20-cycle signal arriving at the repeater from the West line sees two paths, the repeater and the low-frequency branch. The signal does not pass through the repeater, but passes through the low-frequency branch, and thence to the East line. In this way, the ringing signal travels over the whole length of the repeater system. The 20-cycle ringing signal traveling in the opposite direction bypasses the repeater in the same manner.

b. Limitation. This method has the limitation that the ringing signal is attenuated all along the line without any provision for its amplification. The voice-frequency ringing method, which amplifies the ringing signal, is more desirable.

67. Voice-Frequency Ringing

In *voice-frequency ringing,* the 20-cycle ringing signal is converted to a voice-frequency signal by means of a *ringer* located at a point between the near-end switchboard and the near-end terminal repeater; the voice-frequency signal is transmitted through all the repeaters of the line; then by means of a second ringer located between the far-end terminal repeater and the far-end switchboard, the voice-frequency signal is reconverted to 20 cycles. Additional equipment for the frequency conversion is needed only at the terminals. Strictly, the 20-cycle ringing signal is not converted directly into a voice-frequency signal, and vice versa. What actually happens is that the equipment, upon receiving one signal frequency, sends out the other.

a. Circuit. Figure 93 shows the circuit of a voice-frequency ringer. It consists of two major components—the low-frequency receiving circuit and the high-frequency receiving circuit. The low-frequency receiving circuit consists of the low-frequency branch, relay B, and the 1,000/20-cycle source. The 1,000/20-cycle source is an oscillator that puts out a 1,000-cycle signal interrupted at a 20-cycle rate. The high-frequency receiving circuit consists of the high-frequency branch, relay A, and the 20-cycle source. In the figure, the contacts of relays A and B are in the nonoperated position, as they are during conversation; that is, no 20-cycle signal is present at the terminals to the

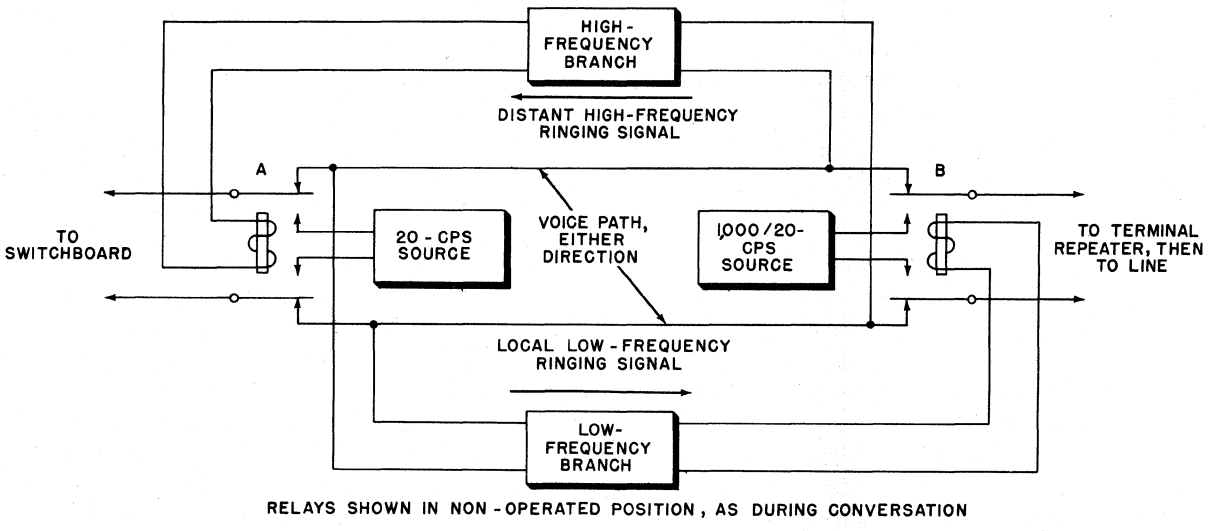


Figure 93. Circuit of voice-frequency ringer.

switchboard and no 1,000/20-cycle signal is present at the terminals to the line.

b. Operation of Low-Frequency Receiving Circuit of Ringer. When a 20-cycle ringing signal enters the ringer from the near-end switchboard (fig.93) it sees two paths—the low-frequency branch and the high-frequency branch. The high-frequency branch will not pass the low-frequency signal, and therefore relay A does not operate. The low-frequency branch does pass the signal, and relay B operates, connecting the 1,000/20-cycle source to the near-end terminal repeater, and then to the line. Thus a voice-frequency signal of 1,000 cycles interrupted at a 20-cycle rate is sent to the other end of the line, where it passes through the far-end terminal repeater into a similar voice-frequency ringer.

c. Operation of High-Frequency Receiving Circuit of Ringer. Assume that the voice-frequency ringer of figure 93 is at the *far-end* switchboard. The 1,000/20-cycle from the line enters the terminal repeater, and then the ringer. The signal sees two paths to the ringer—the low-frequency branch and the high-frequency branch. The low-frequency branch will not pass the high-frequency signal, and consequently relay B does not operate. The high-frequency branch does pass the signal, and relay A operates, connecting the 20-cycle source to the switchboard. Thus, a 20-cycle ringing signal is sent to the switchboard. The combination of a 1,000-cycle signal interrupted at a 20-cycle rate is used to prevent false operation of relay A from voice signals during conversation. The high-frequency branch rejects all frequencies except the single combination of a 1,000-cycle signal interrupted at a 20-cycle rate. The probability of such a frequency combination occurring during a conversation and persisting long enough to cause operation of relay A is very remote.

d. Use of Other Ringing Frequencies. Although a voice-frequency ringing source of 1,000/20 cycles is used in the explanation in *c* above, it is not the only voice frequency used in voice-frequency ringers. Some use 1,225 or 1,600 cycles. These employ time-delay circuits, so that a sustained

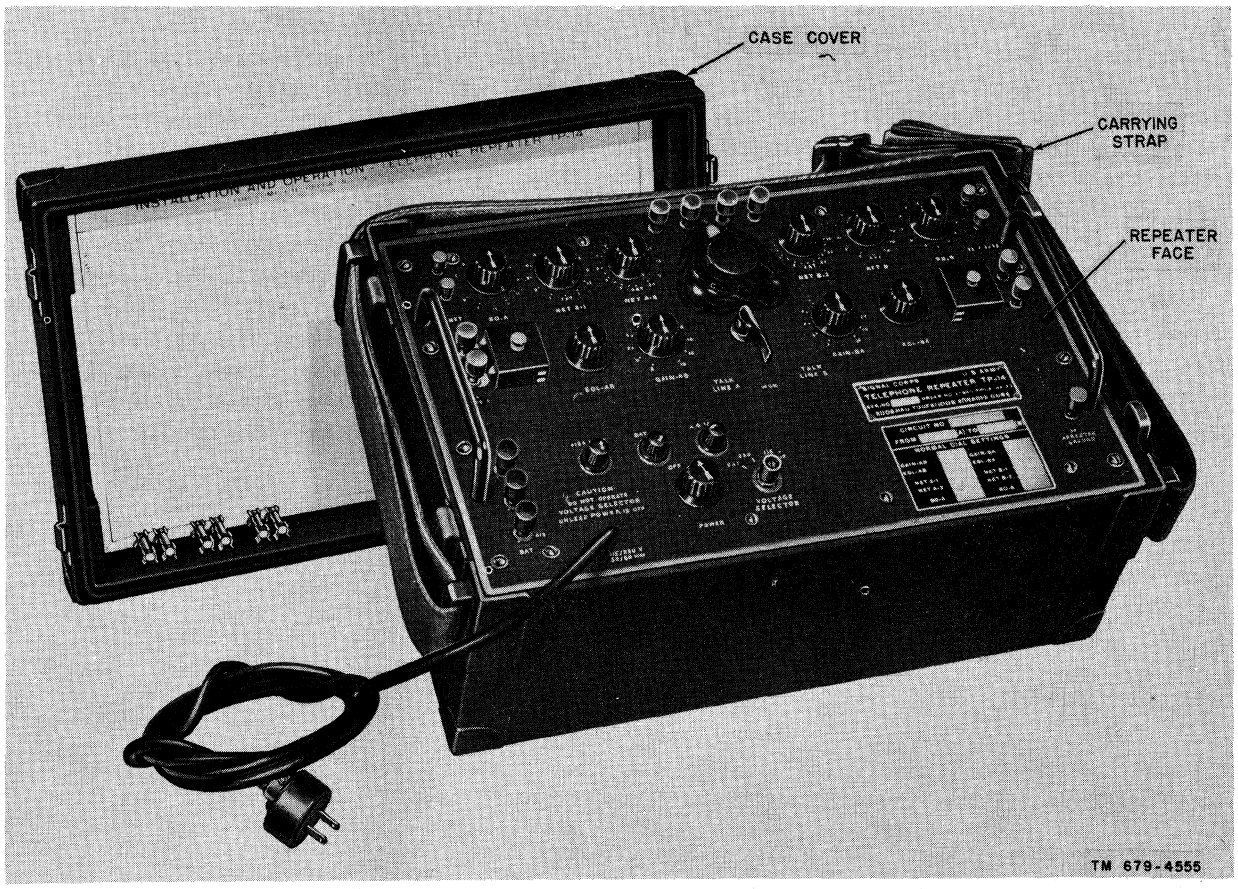


Figure 94. Telephone Repeater TP–14.

signal of 1,225 or 1,600 cycles—whichever is used—is needed to operate the ringer.

68. Army Hybrid Arrangements and Repeaters

The Army uses various types of hybrid arrangements and repeaters; chief among them are the ones described briefly in *a* through *e* below.

a. One voice-frequency hybrid is a portable unit which covers the voice-frequency range from 250 to 3,200 cycles. It uses a universal adjustable balancing network. A four-wire impedance matching circuit accomodates the nominal impedance values of a majority of four-wire equipments. These values are 150, 600, and 4,000 ohms for four-wire incoming (4W–IN) connections and 150 and 600 ohms for four-wire outgoing (4W–OUT) connections. The unit contains a remote-control system, used when radio sets are attached to the four-wire side.

b. Figure 94 shows a portable 22-type voice-frequency repeater for tactical use. It is provided with a removable case cover and a carrying strap. On the panel are mounted binding posts, used for all external connections, and variable setting dials which permit independent variation of the repeater gain for both directions of transmission. The unit contains two built-in adjustable balancing networks designed to permit operation on most types of Army lines. It also contains a low-frequency bypass circuit for bypassing 20-cycle ringing current around the repeater, when desired. It operates at from 115 or 130 volts, 50 to 60 cycles, or at 12 volts direct current (battery supply). This repeater is approximately 19 inches wide.

c. Figure 95 shows a portable 21-type voice-frequency repeater for tactical use (principally on nonloaded field wires). It is provided with a hinged case cover and a carrying strap. On the panel are mounted binding posts, used for all external connections, and one variable setting dial. Since it is a 21-type voice-frequency repeater, it contains only 1 amplifier for both directions of transmission and only 1 dial for variation of amplifier gain. It operates on a dry-cell battery which is connected to the amplifier section by means of the battery cable. This repeater is approximately 6 inches wide.

d. Figure 96 shows a voice-frequency repeater employed in fixed plant installations. It is used as a 22-type, a 44-type, or a terminal-type repeater on open-wire or cable facilities. Note the location of the several panels. Each is a separately mounted unit. The bridging and telephone-set panel is used by the attendant for monitoring. The repeater panel consists of two single-stage amplifiers which operate in opposite directions. The balancing-network panel consists of two adjustable-network circuits for balancing the line circuits in both directions from a two-wire repeater. The voice-frequency ringer panel consists of two voice-frequency ringers. The power-supply panel operates at from 105 to 125 volts, 50 to 60 cycles.

e. Figure 97 shows a voice-frequency repeater, basically the same as the repeater described in *d* above. It differs in that it contains 3 repeater panels, 3 balancing network panels, and 2 power-supply panels, but no voice-frequency-ringer panel. Each of these multiple panels is identical with the corresponding panel in figure 96. There are as many balancing-network panels as repeater panels because each repeater requires a balancing network. There are 2 power-supply panels because 3 repeaters require that many for their operation.

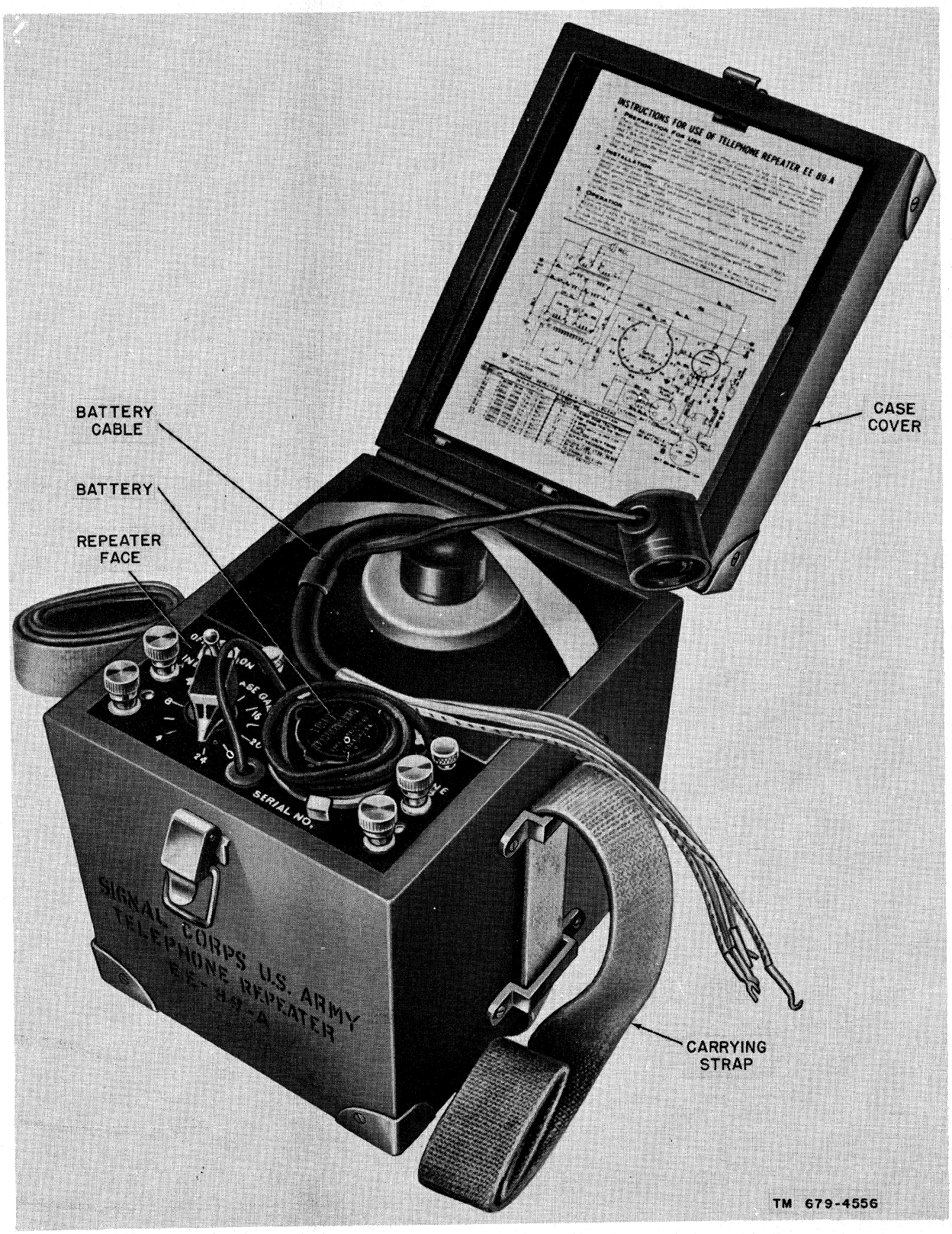


Figure 95. Telephone Repeater EE–89–A.

282463 O—54——6

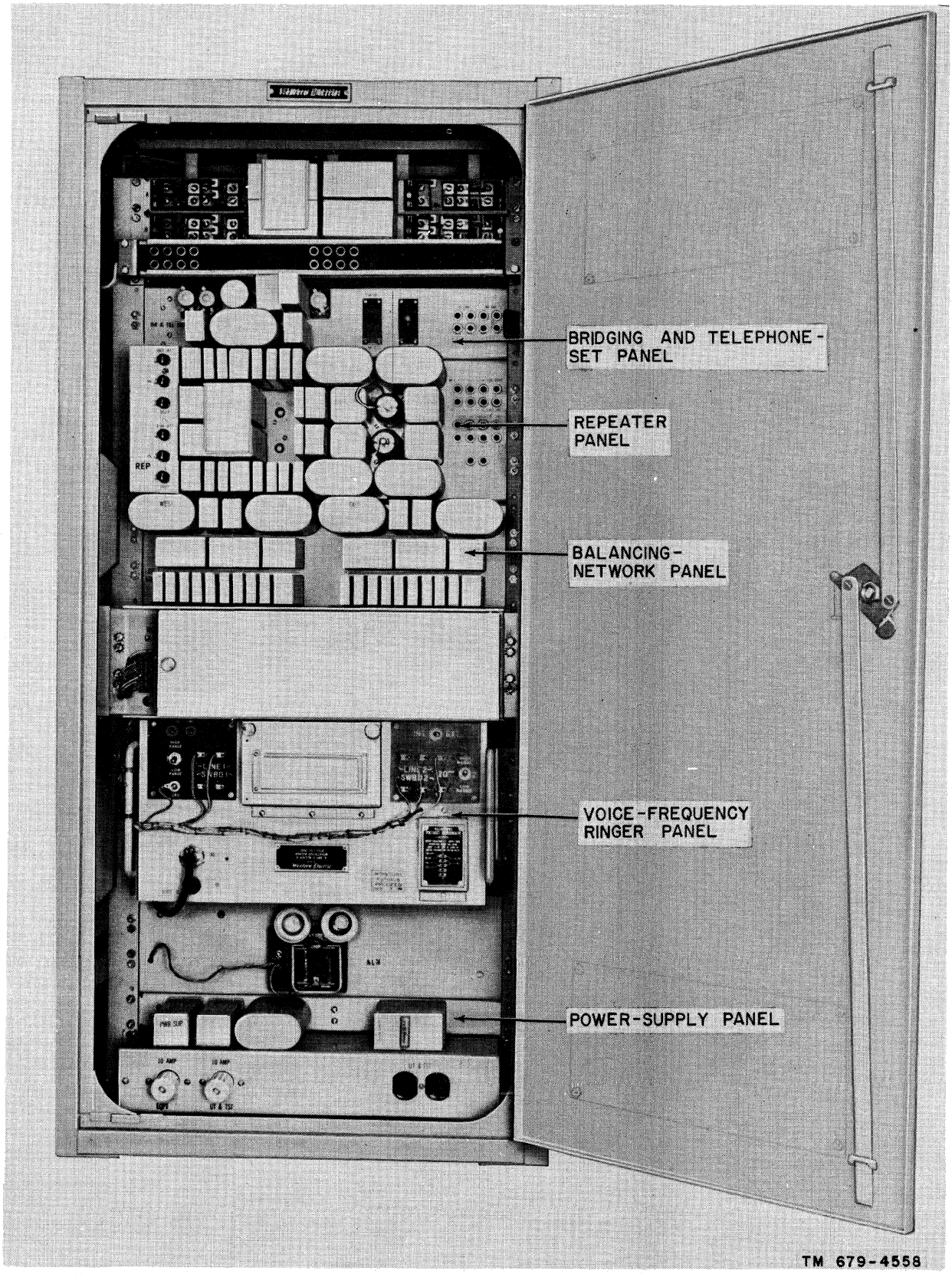


Figure 96. Telephone repeater.

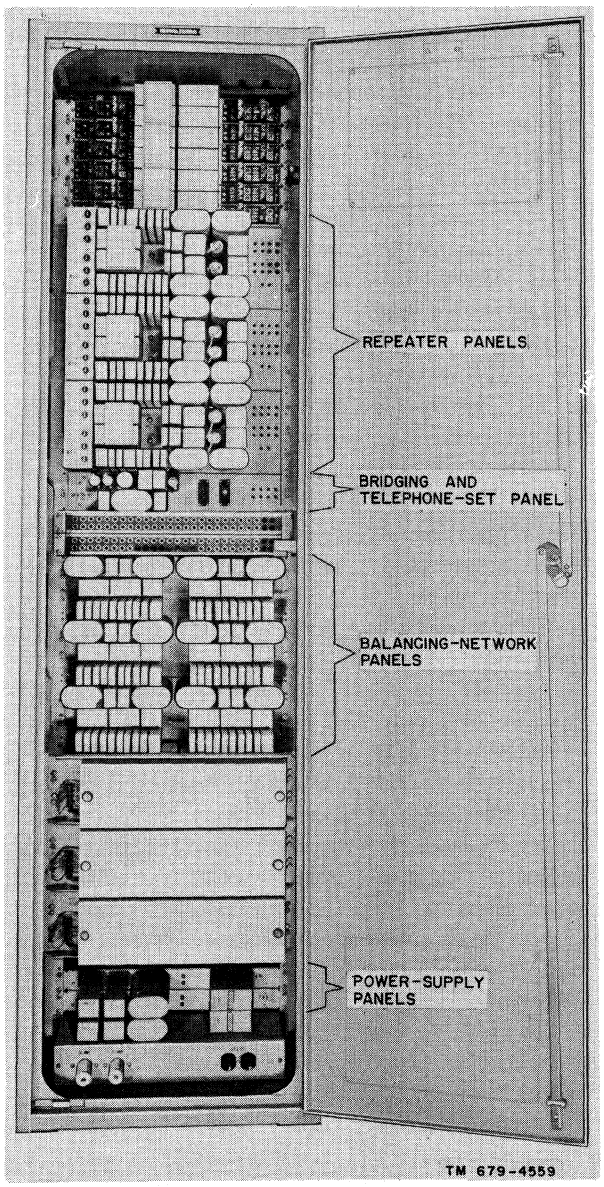


Figure 97. Telephone repeater.

69. Summary

a. One of the problems of long transmission lines is the serious loss of power. Power equal to the power lost can be put back into the lines, however, by use of electronic amplifiers.

b. The electronic amplifier is a one-way device.

c. The combination of amplifier or amplifiers with attached networks is called a repeater.

d. A repeater that terminates a repeater system is called a terminal repeater. Any other is called an intermediate repeater.

e. Power-level diagrams are used to show the signal strength at every point in a transmission-line system.

f. When two amplifiers of a repeater are connected so that the output of one is fed directly to the input of the other, a complete feedback loop results. Unwanted random oscillations occur, and voice frequencies, among others, may be generated. Such oscillation is called singing, and the repeater is said to sing.

g. To eliminate singing, it is necessary to separate the sending and receiving paths at the junction of a two-wire and a four-wire system. The network that accomplishes this is called a hybrid set or hybrid arrangement.

h. Four types of hybrid arrangements are the hybrid coil, the repeating-coil hybrid, the four-wire terminating set, and the resistance hybrid.

i. The theory of operation of hybrid arrangements is based upon one or more of the following circuits—the d-c Wheatstone bridge, the a-c Wheatstone bridge, and transformer coupling.

j. The equation at balance for the d-c Wheatstone bridge is $\frac{R1}{R2}=\frac{R3}{R4}$, where $R1$ and $R2$ are adjacent arms.

k. The equation at balance for the a-c Wheatstone bridge is $\frac{Z1}{Z2}=\frac{Z3}{Z4}$.

l. Transformer coupling is used for impedance matching and the electrical isolation of circuits. Maximum power transfer is obtained when impedances are matched.

m. Hybrid action is the separation of the sending and receiving paths at the junction of a two-wire and a four-wire system. A modified impedance bridge, when balanced, can be used to separate the sending and receiving paths.

n. The network attached to a hybrid arrangement for the purpose of balancing the transmission line is called the balancing impedance or, more commonly, the balancing network.

o. The hybrid coil, repeating-coil hybrid, and four-wire terminating set use transformer coupling to obtain hybrid action.

p. The hybrid coil is a single transformer having three windings, used to separate the sending and receiving paths at the junction of a two-wire and a four-wire system.

q. A 173-type repeating coil consists of six separate windings each of which is magnetically coupled to all the others. Two 173-type repeating coils, when connected to provide a hybrid arrangement, are called a repeating-coil hybrid.

r. The rule to be followed when considering the polarity of the induced voltages in the windings of a repeating coil is that odd-numbered terminals have the same polarity, and even-numbered terminals have the opposite polarity. It follows that the odd-numbered and even-numbered terminals have opposite polarities.

s. Opposing magnetic fields induce opposing voltages in a winding or a pair of windings.

t. The strength of a magnetic field depends on the product of the number of turns of winding and the current in the winding.

u. A four-wire terminating set is a hybrid arrangement of 2 repeating coils having 4 windings in each coil.

v. Because ideal coils cannot be manufactured a very small signal does reach the input of one amplifier from the output of the other. This very small amount is tolerable, but it limits the amount of amplification that can be provided by a repeater without singing.

w. A symmetrical hybrid arrangement is one in which the transmission loss in *db* is the same for both directions of transmission.

x. As applied to hybrid arrangements, transmission loss is the loss in signal power through hybrid action plus coil-core and winding-resistance losses. The loss for the transformer-type hybrid arrangements, in either direction, is approximately 3.25 to 4.0 *db*.

y. In a repeater, the loss of signal power from the output of one amplifier to the input of the other amplifier is called the transhybrid loss.

z. The power not reflected is called the return loss (*R. L.*).

aa. The transhybrid loss is equal to the sum of the return loss and twice the transmission loss.

ab. Various types of balancing networks such as compromise networks, line-balancing networks, building-out networks, and line-equipment balancing networks are provided in a number of standard arrangements.

ac. A resistance hybrid consists of four equal center-tapped resistors. It does not give as precise a balance as the hybrid coil or the repeating-coil hybrid.

ad. The transmission loss of a resistance hybrid is approximately 9.5 *db*.

ae. Four common types of repeaters are the 21-type, 22-type, terminal-type, and 44-types.

af. An amplifier consists of 1 or more stages of amplification—that is, 1 vacuum tube, or more than 1 vacuum tube connected in cascade, with a coupling network or networks.

ag. The 21-type repeater contains a single amplifier and a single hybrid coil, and has a circuit arrangement which requires no balancing network It is used in a two-wire system.

ah. The 22-type repeater contains 2 amplifiers and 2 hybrid arrangements with balancing networks. It is used in a two-wire system.

ai. When the total gain in the circuit between the amplifiers of a repeater equals or exceeds the total loss of that circuit, sufficient feedback exists to cause singing.

aj. Barreling is a near-singing condition, harmful to transmission.

ak. The difference between the total loss in the singing path and the total gain of the amplifiers, when set for normal operation, is called the singing margin. A singing margin of 8 *db* is the minimem for satisfactory operation.

al. Passive singing-point tests permit measurement of the transhybrid loss of a single hybrid arrangement.

am. Active singing-point tests permit determination of the actual operating conditions of a 22-type repeater system.

an. The 44-type repeater employs two amplifiers and no hybrid arrangements. It is used in a four-wire system.

ao. The terminal-type repeater employs 2 amplifiers and only 1 hybrid arrangement. It is used at the termination of a four-wire repeater system.

ap. Selection of the repeaters for a system and determination of their spacing in the line require knowledge of the amplification capabilities of the repeaters and the characteristics of the line.

aq. Antihybrid devices are devices which eliminate the need for hybrid arrangements in a two-wire voice-frequency repeater system.

ar. Anti-hybrid devices are used in push-to-talk operation and voice operation.

as. Two methods of ringing over a voice-frequency repeater system are voice-frequency ringing and 20-cycle ringing.

70. Review Questions

a. What is the lowest allowable power level in *dbm* at the telephone receiver, for good military communication?

b. What device is used to compensate for line loss?

c. Is an amplifier a one-way device?

d. What is a repeater?

e. Where is a terminal repeater located?

f. Where is an intermediate repeater located?

g. What is the purpose of a power-level diagram?

h. What is meant when a repeater is said to sing?

i. What is the purpose of a hybrid arrangement?

j. Name four types of hybrid arrangements.

k. It is desired to build a balanced d-c Wheatstone bridge using two 1,000-ohm and two 200-ohm resistors. Draw a schematic diagram showing the resistors, the galvanometer, and the battery.

l. Name two uses for transformer coupling.

m. What is the hybrid coil?

n. Of what does the repeating-coil hybrid consist?

o. Of what does the four-wire terminating set consist?

p. Where does the signal power go that enters a hybrid arrangement from the output of the East–West amplifier? Of the West–East amplifier?

q. Where does the signal power go that enters a hybrid arrangement from the attached transmission line?

r. What is meant by a symmetrical hybrid?

s. What is transmission loss as applied to hybrid arrangements?

t. Describe return loss.

u. What is transhybrid loss?

v. What is the significance of a large transhybrid loss?

w. What is the purpose of a balancing network?

x. Are there standard arrangements of balancing networks?

y. Are compromise networks used when a high degree of balance is desired?

z. What is a complete network?

aa. Of what does a resistance hybrid consist?

ab. Does the resistance hybrid have a smaller or a greater transmission loss than the repeating-coil hybrid?

ac. Which two hybrids give a higher degree of balance?

ad. Draw a simplified schematic for each of the four types of repeaters.

ae. What is the purpose of a passive singing-point test?

af. What is the purpose of an active singing-point test?

ag. What must be the relationship between the total gain and the total loss in the singing path in order that a repeater will not sing?

ah. What is barreling, and when is barreling present?

ai. What is singing margin?

aj. What is the minimum singing-margin value for satisfactory operation?

ak. What is the purpose of an antihybrid device?

al. Why is it not possible for a 20-cycle ringing signal to take the same path as voice frequencies over a voice-frequency repeater system?

am. Briefly describe two methods of ringing over a voice-frequency repeater system.

CHAPTER 4
CARRIER TELEPHONY

71. Introduction

a. Purpose of Carrier Systems.

(1) Theoretically, a transmission line is capable of transmitting frequencies ranging from zero to infinity. For practical considerations, however, an open-wire transmission line is capable of efficiently transmitting frequencies ranging only from zero to approximately 150,000 cycles. Since the voice-frequency range can be considered to be from 200 to 2,800 cycles (a range of 2,600 cycles), only a very small portion of the potential frequency-range capabilities of the line is utilized when a single voice conversation is transmitted over an open-wire transmission line. In effect, the entire frequency range from 2,800 to approximately 150,000 cycles is wasted.

(2) There is, however, a method of *reallocating* (spacing out) several 2,600-cycle normal voice channels over this 2,800- to 150,000-cycle range, which otherwise would be wasted. This method makes it possible to transmit several conversations simultaneously without interference, and thereby utilize the transmission line fully. Also, it requires fewer transmission lines to carry a given number of simultaneous conversations between switchboards, so that the personnel required for line construction and maintenance are greatly reduced. This reallocation, to obtain several facilities over the same line, is accomplished by *carrier systems*, and is known as *frequency division multiplexing*.

b. Nature of Transmitted Signal.

(1) In a carrier system, an alternating current of a constant, predetermined frequency is generated at the terminal of the carrier system by an *oscillator*. This current is called the *carrier current*, because it serves to *carry* the voice-current intelligence which is superimposed on it. The process of superimposing the voice-current intelligence on the carrier current is called *modulation*, and the device which makes this process possible is called a *modulator*. One of the results of impressing the voice-frequency intelligence on the carrier frequency is a reallocation, or shifting, of the voice-frequency range.

(2) For example, the 2,600-cycle band of voice frequencies of from 200 to 2,800 cycles can be shifted by a carrier current of suitable frequency to occupy a band of from 3,100 to 5,700 cycles; therefore, it is possible to transmit two messages simultaneously on the same transmission line without mutual interference, one message occupying a 2,600-cycle range of from 200 to 2,800 cycles, the other occupying a 2,600-cycle range of from 3,100 to 5,700 cycles. If several different carrier frequencies separated sufficiently in frequency are produced, so that the message currents do not interfere with each other, several messages can be transmitted over the same wire facility just as independently as if there were separate lines for each one.

(3) At the receiving end, the voice intelligence is separated from the carrier by a process called *demodulation*, and the device which makes this process possible is called a *demodulator*. The process of demodulation is similar to that of modulation, by which the voice-current intelligence is superimposed on the carrier current.

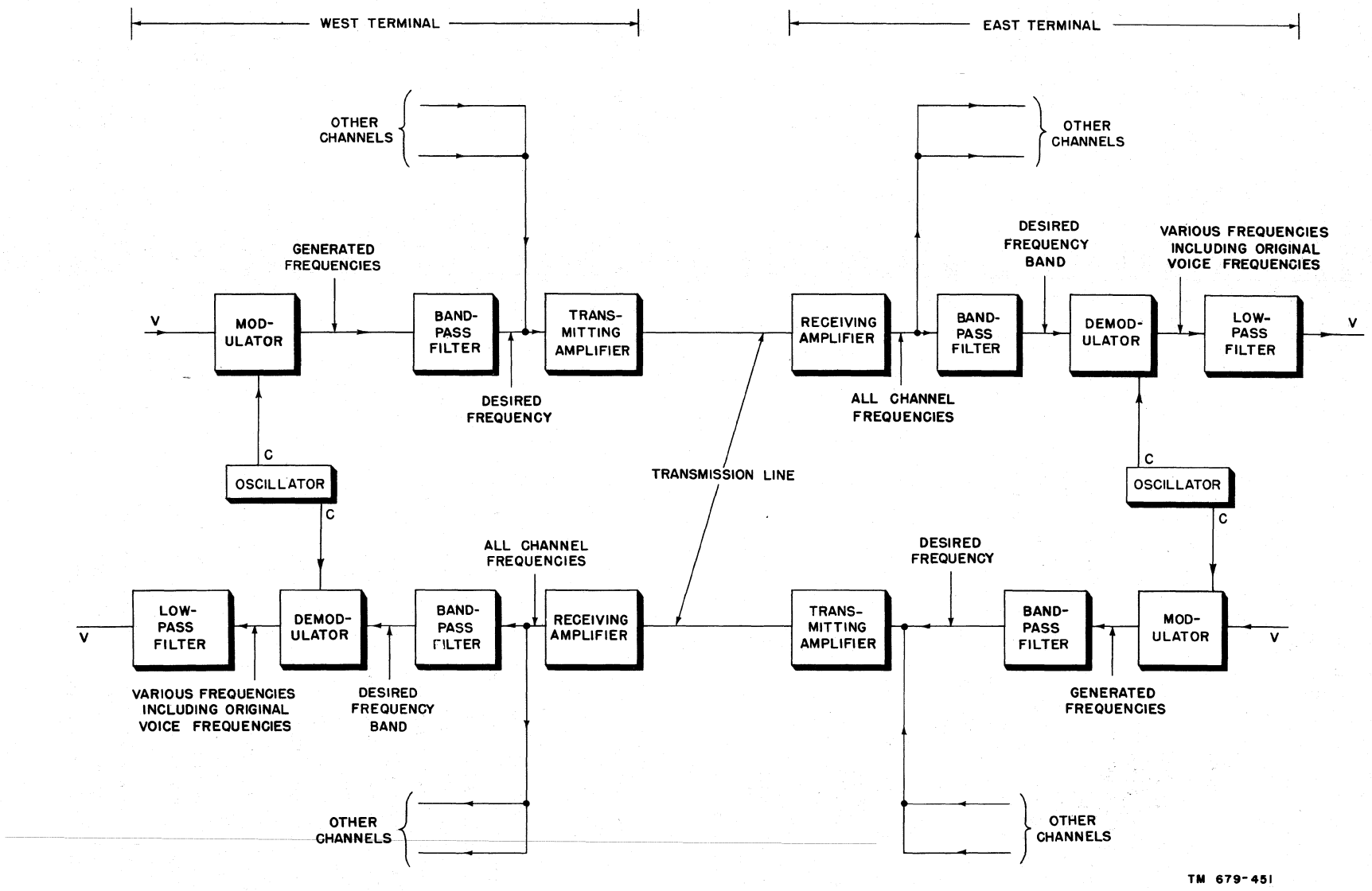


Figure 98. Basic carrier system.

72. Basic Elements of Carrier System

Figure 98 is a block diagram of a *basic carrier system.* This can be said to consist of three major components: the West terminal, the transmission line, and the East terminal. Each terminal is made up of several *channels,* each of which provides for a single two-way conversation. Only the basic elements of a single channel are shown.

a. Trace the path for transmission of voice signals from one terminal to the other, starting at the West terminal (fig. 98). Voice signals, V, enter the *modulator* from an adjacent switchboard, and the carrier current, C, enters the modulator from an *oscillator.* The output of the modulator consists of various frequency conponents produced in the modulator, the particular ones produced depending on the type of modulator used. Not all the frequency components produced carry the voice intelligence. It is necessary to transmit only one of the voice-intelligence components; therefore, transmission of only one certain frequency component, of all the frequency components produced, is desired. (If all of these frequencies were transmitted, large frequency spacing would be needed between channels to prevent interference—which would reduce the number of channels that could be used.)

b. To suppress the undesired frequencies and permit ready passage of the desired frequency range, there is need for a device to discriminate between frequencies. Such a device is called a *band-pass filter* (fig. 98). It consists of capacitive and inductive reactances. The band-pass filter in each channel permits ready passage of the frequency range of that channel, and all the channels feed their output frequencies into a common transmitting amplifier. From there, all these frequencies are sent out into the transmission line.

c. At the East terminal, the frequencies enter a common receiving amplifier, as shown. At the amplifier output, other *band-pass filters* in the several channels select the desired frequency. The desired frequency carrying the voice intelligence, and the output of an *oscillator* equal in frequency to the original oscillator frequency are applied to the demodulator. The output of the demodulator consists of various frequency components produced in the demodulator. One of these components is the original voice frequency, V.

d. The *low-pass filter* that follows permits only the voice frequencies, V, to pass readily to the receiving switchboard (fig. 98). The low-pass filter is similar to the band-pass filter in that it consists of capacitive and inductive reactances, and it discriminates between frequencies; however, it passes frequencies from nearly zero up to a predesignated frequency (usually about 2,800 cycles). Such a filter is adequate at this point, since all the other frequency components are greater than the original voice frequency.

e. The process for transmission in the opposite direction is the same.

f. The modulator and demodulator in the transmit and receive circuits of a carrier channel usually derive their carrier frequency from a common source, as shown. The term *modem* is used to indicate the modulation and demodulation function in the carrier channel.

73. Modulation

When two frequencies are combined in a modulator in the process called *modulation,* one or both of the original frequencies appear in the output together with several other frequencies. Two of these other frequencies of particular concern are the *sum* of the two original frequencies, called the *sum frequency,* and the *difference* between the two original frequencies, called the *difference frequency.*

a. Figure 99 is a simplified block diagram of a modulator followed by a band-pass filter. The two frequency components that are combined in the modulator are a voice-frequency band of 200 to 2,800 cycles, V, and a carrier frequency of 6,000 cycles, C. The output frequency components of particular concern are the sum-frequency band of 6,200 to 8,800 cycles (C+V) and the difference-frequency of 3,200 to 5,800 cycles (C—V). The original carrier frequency may or may not appear in the output, depending on the type of modulator used. A balanced-bridge modulator (par. 79) suppresses the carrier.

b. The sum-frequency band and difference-frequency band also are called, respectively, the *upper-modulation component* and *lower-modulation component;* or, more commonly, the *upper sideband* and the *lower sideband.* Each of these frequency components contains the voice-frequency intelligence in such a way that the demodulator can reproduce the original voice-frequency band; therefore, only one sideband, either the upper or the lower, need be used to convey the message (fig. 99). The band-pass filter, by its design, permits only one of the sidebands to pass readily—either the 6,200- to 8,800-cycle (C+V) sideband

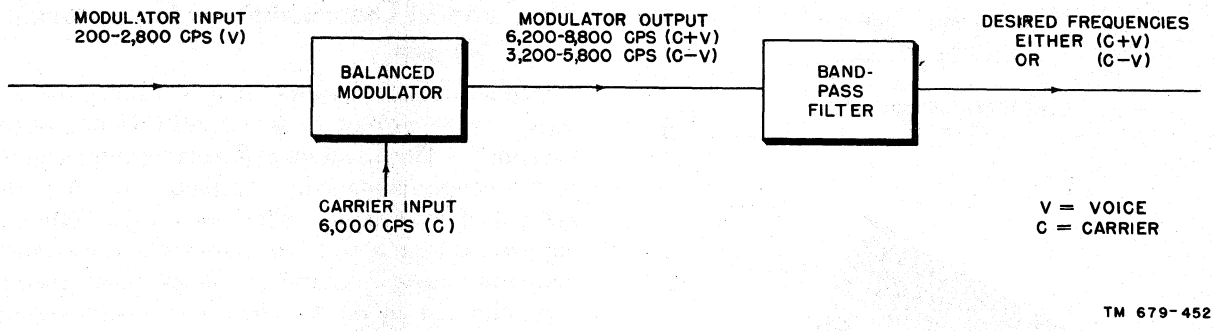


Figure 99. Modulation and transmission of one sideband.

or the 3,200- to 5,800-cycle (C—V) sideband; all of the other frequencies are suppressed. This is known as *single-sideband transmission with suppressed carrier*, and is used in military carrier equipment.

74. Demodulation

In a sense, *demodulation* is remodulation, for in demodulation the sideband reaching the *demodulator* is combined with the frequency of the original carrier to reconstruct the original voice-transmitted signal.

a. The combination of frequency components in the output of a demodulator is the same as in the output of a modulator; that is, one or both of the frequencies that are combined in the demodulator appear at the output of the demodulator together with several other frequencies. Two of the other frequencies are the sum and difference of the input frequencies. The frequency of particular concern is the *difference frequency*. The circuit arrangement of a demodulator can be the same as that of a modulator.

b. Assume that the band-pass filter of figure 99 passes only the lower sideband of 3,200 to 5,800 cycles (C—V). This lower sideband is transmitted to the receiving terminal. Figure 100 shows this lower sideband entering the demodulator at the receiving terminal, where it is combined with the 6,000-cycle carrier produced by a local oscillator to provide a difference-frequency band of 200 to 2,800 cycles, (C-(C—V), or V. This difference-frequency band, as has been said, is the original voice-frequency band. A sum-frequency band of 9,200 to 11,800 cycles (C+(C—V)), or 2C—V, also is produced, as well as other output frequency components, but the low-pass filter that follows the demodulator permits only the voice-frequency band to pass readily to the adjacent switchboard. If the upper sideband of 6,200 to 8,800 cycles, (C+V), were transmitted, the difference-frequency band in the output of the demodulator again would be the original voice-frequency band of 200 to 2,800 cycles. Hence, in demodulation, the original voice-frequency band is reconstructed by combining the frequency of the original carrier with whichever one of the side bands that is received.

75. Copper-Oxide Varistor

a. Use of Copper-Oxide Varistor. *Copper-oxide* rectifiers are widely employed in the modulators and demodulators (modems) of present-day carrier systems. These rectifiers commonly are called *copper-oxide varistors*, or simply *varistors*. Vacuum tubes may be used for rectification, modulation,

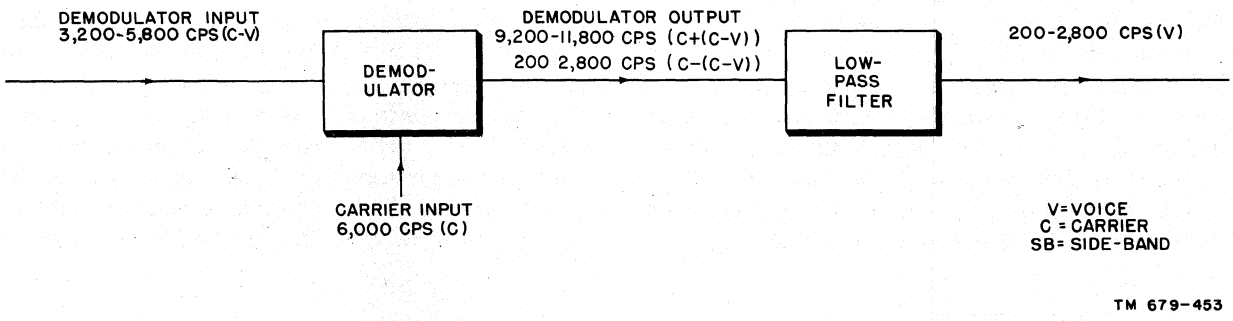


Figure 100. Demodulation and transmission of voice signal.

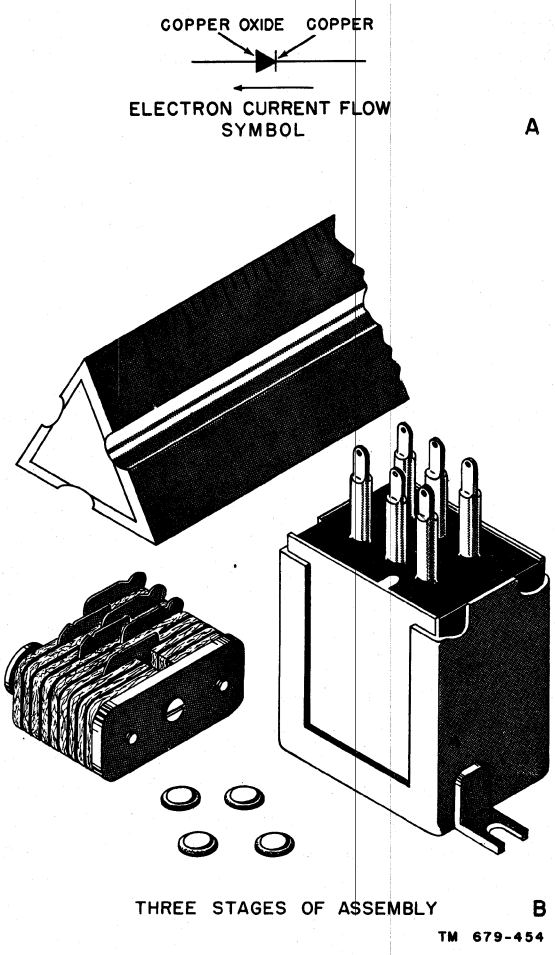


Figure 101. Copper-oxide varistor.

and demodulation in carrier systems, but varistors usually are given preference because of their simplicity, greater stability, and apparently unlimited life.

b. Structure of Copper-Oxide Varistor. B, figure 101, shows a copper-oxide varistor in three stages of assembly. In the foreground are four copper-oxide disks of 3/16-inch diameter. In the back are the disks assembled with their connections, and the complete assembly in its can. Note the relative sizes as compared to the scale. Each disk is a small perforated copper plate on one side of which is a layer of copper oxide, obtained by heating the plate to a very high temperature in air. Ordinary means serve to connect an external circuit to the copper side, but a special vaporized-metal contact is necessary for the oxide side.

76. Electrical Characteristics of Copper-oxide Varistor

Copper-oxide varistors, like vacuum-tube diodes, permit current to flow readily in one direction only. The direction of flow is from the copper to the copper-oxide side. In figure 101, A is the symbol of a varistor. The *wedge* represents the copper oxide and the *line* represents the copper. Electron current, therefore, flows most readily from the line to the wedge. The wedge symbol direction is set up to indicate the direction of conventional current. In this manual, electron current is used, and flows in the opposite direction to conventional current.

a. In A, figure 102, the current source is a battery placed across a copper-oxide varistor. The battery causes the copper to be negative with respect to the copper oxide, and, as a result, there is an appreciable flow of electron current in the direction as shown. This means that the varistor presents a small resistance. The amount of resistance depends on the amount of voltage across the varistor. In B, the polarity of the battery is reversed. There is a minute flow of current, but it is so slight that for all practical purposes it may be said that there is no flow at all. This means that the varistor now presents a very large resistance. Figure 103 is an experimentally obtained plot showing the variation of varistor resistance with potential. It shows that when the copper is negative with respect to the copper oxide, the resistance of the varistor decreases to a fairly constant value as the amount of potential increases. When the copper is positive with respect to the copper oxide, the resistance increases to a very large value, and then decreases somewhat as the potential increases.

b. To better understand the great change in resistance when the potential polarity is reversed, consider this example. When +1.2 volts is placed across the varistor (copper oxide positive with respect to copper), the resistance is 32 ohms (fig. 103). When −1.2 volts is placed across the varistor (copper oxide negative with respect to copper), the resistance is 85,000 ohms. Consequently, +1.2 volts across the varistor has a current flow approximately 2,700 times as great as −1.2 volts across it. For this reason, the current flow is said to be zero (it is negligible) when there

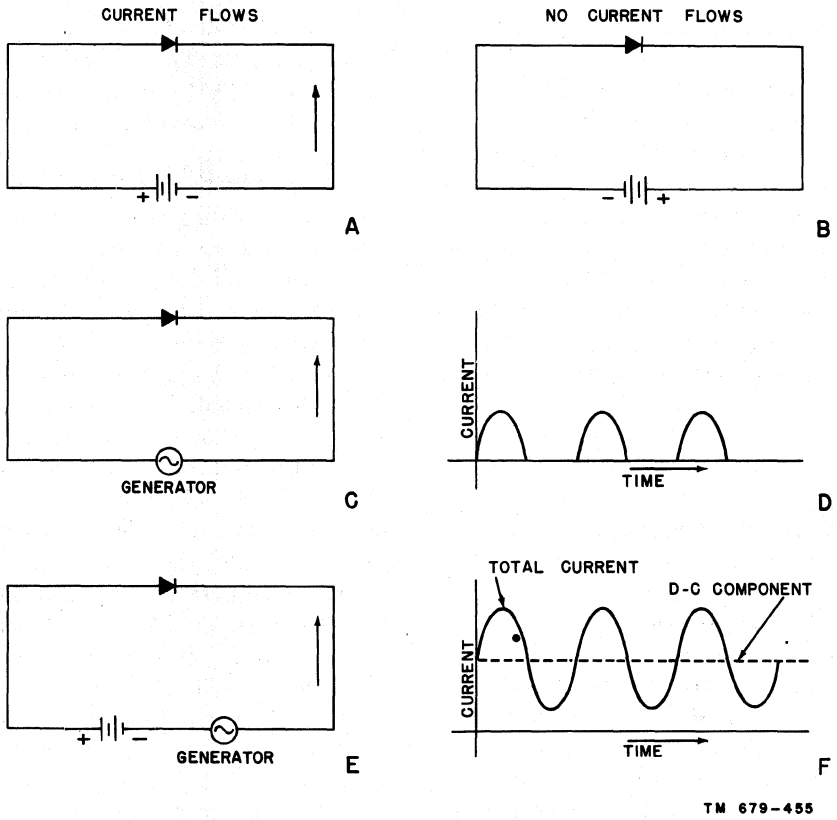


Figure 102. Characteristics of copper-oxide varistor.

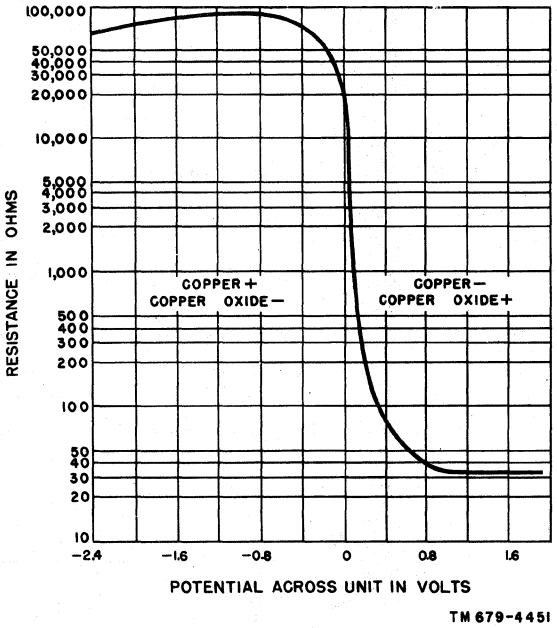


Figure 103. Resistance variation of copper-oxide varistor.

is a negative polarity across the varistor; that is, when the copper oxide is negative with respect to the copper. When current from an a-c generator is applied to a varistor (C, fig. 102), the voltage polarity across the varistor is positive for one half-cycle of the applied generator voltage and negative for the other half-cycle. Therefore, current flows only during the positive half-cycle (D, fig. 102). This is known as half-wave rectification.

c. E shows a battery in series with an a-c generator across a copper-oxide varistor. Assume that the battery voltage is much greater than the peak a-c voltage. The polarity across the varistor, then, is always such as to make the copper negative and the copper oxide positive with respect to each other, and current always flows. The magnitude of this current always is changing, however, because the total voltage across the varistor is changing.

d. F, figure 102, shows how the current varies. The dotted line indicates the current of the battery only; the total current is this d-c current plus the a-c component from the generator. Note that the larger d-c voltage permits this alternating current to pass through the varistors without being rectified; that is, if the d-c voltage exceeds the peak a-c voltage, and is applied in the polarity shown in

E, the varistor will have a small resistance at all times.

e. Assume that the battery polarity shown in E is reversed. Since the battery voltage is much greater than the peak value of a-c voltage, the copper is now always positive and the copper-oxide always negative, and no current flows. Used this way, the larger d-c voltage does not permit a-c current caused by a smaller a-c voltage to pass through the varistor; that is, the large d-c voltage causes the varistor to have such a very large resistance that for all practical purposes there is an open circuit.

f. It is evident that a d-c voltage can be used to control the resistance value of a varistor. Since the resistance changes so greatly when the polarity changes, a varistor can be considered an electrical switch that is controlled by the d-c voltage.

77. Germanium-Crystal Diode

In many applications, *germanium-crystal diodes* are replacing copper-oxide rectifiers and vacuum-tube diodes. A crystal diode consists essentially of a sharply pointed wire in contact with a small slab of semiconducting material. In the germanium-crystal diode, the sharply pointed wire is a fine tungsten whisker formed with a precision point, and the semiconductor is germanium. Germanium diodes have electrical characteristics similar to those of copper-oxide rectifiers, offering low resistance to current flow in one direction and very high resistance in the other. Electron current flows readily from the germanium to the tungsten, but flows practically not at all in the reverse direction. The smaller size of the germanium-crystal diode is an advantage. The symbol for a crystal diode is the same as that for a copper-oxide rectifier. The wedge represents the tungsten, and the line represents the germanium.

78. Modulators and Demodulators

a. Telephone carrier terminals tend to become much more complex as more and more channels are placed on a single pair of wires. Against this tendency, the extreme simplicity and reliability of varistor, type and germanium-crystal diode modulators and demodulators are of great value. The swing toward use of germanium-crystal diodes in modulators and demodulators is only a recent trend, however.

b. Two common types of modulators and demodulators are the balanced-bridge and lattice types. These circuits are analyzed, primarily from the point of view of the modulator, in paragraphs 79 through 84. The demodulator circuits do not require special attention, since they operate in the same way as modulator circuits.

79. Balanced-Bridge Modulator

In the modulators and demodulators of some carrier-telephone systems, varistor units or germanium-crystal diodes are connected in a Wheatstone-bridge arrangement to obtain the desired modulation or demodulation results.

a. Figure 104 shows a *balanced-bridge modulator.* The voice signal, V, is fed across the terminals that go to the output; the carrier signal, C, is connected across the other two terminals of the bridge. The rectifying units used in a balanced-bridge modulator operate as carrier-controlled switches; therefore, the carrier voltage must be greater than the voice voltage. The rectifying units perform in such a manner that either a short circuit or an open circuit is placed across the output for the voice signal. The voice signal is obtained from the secondary winding of an input transformer, and the output is sent directly to the primary winding of an output transformer.

b. This modulator (fig. 104) combines the voice and carrier frequencies in such a manner that the output contains the original voice frequency, the upper and lower side bands, and other frequencies which are not desired. The original carrier frequency is not in the output, however.

80. Carrier-Frequency Voltage Switching in Balanced-Bridge Modulator

In order to understand the operation of the balanced-bridge modulator, it is necessary to know the exact manner in which the carrier-controlled switches are operated by the carrier voltage.

a. In figure 105, A shows a balanced-bridge modulator to which is applied carrier voltage only. R1 is the input impedance of the voice source, and R2 is the impedance of the modulator load. The

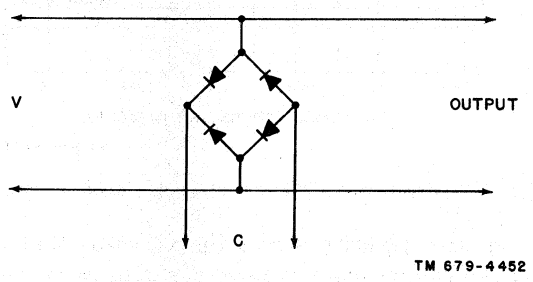


Figure 104. Balanced-bridge modulator.

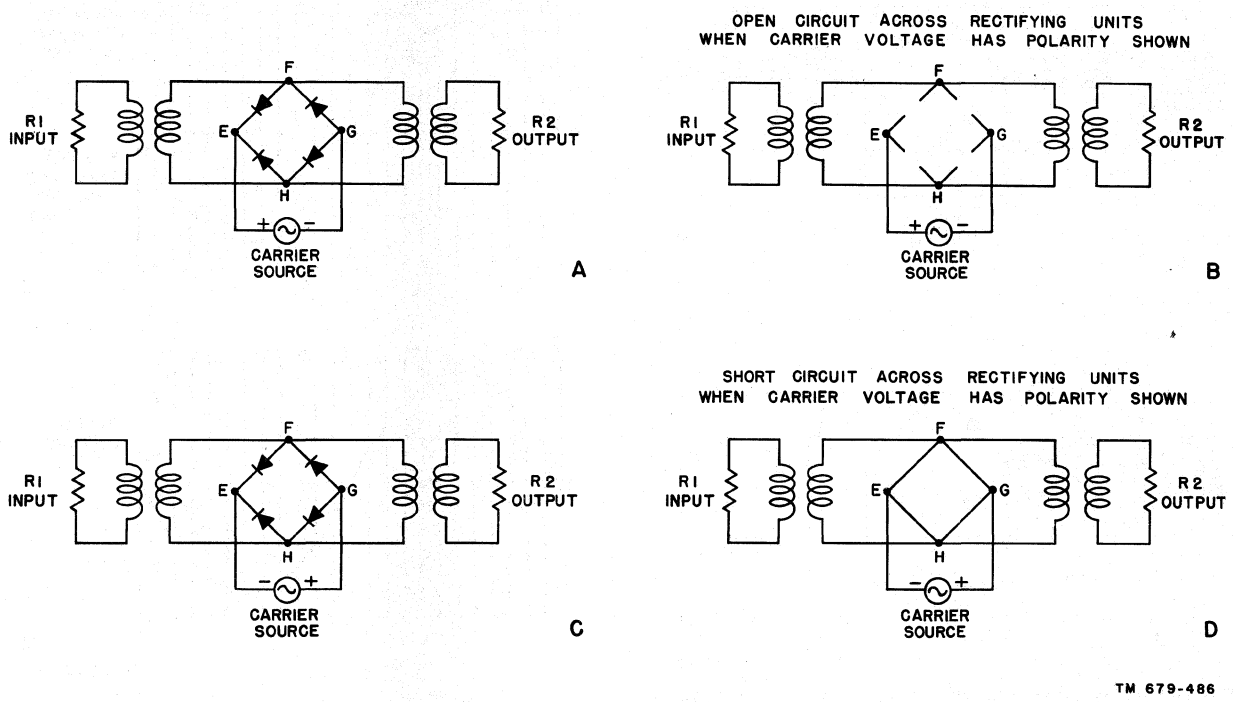


Figure 105. Switching effect of carrier-frequency voltage in balanced-bridge modulator.

carrier source has the polarity indicated; thus, G is negative and E is positive. Notice that no current flows in the rectifying units. (Compare this condition with that of the circuit in B, figure 102.) This means that the rectifying units have a very large value of resistance at this instant, and can be considered as open-circuited. B, figure 105, emphasizes the open-circuit effect of this polarity across the rectifying units.

b. During the next half-cycle, the carrier source changes its polarity, as shown in C, figure 105. Now, G is positive and E is negative, and current flows. (Compare this condition with that of the circuit in A, figure 102.) This means that the rectifying units have a very small value of resistance at this instant and can be considered as short-circuited. D, figure 105, emphasizes the short-circuit effect of this polarity across the rectifying units.

c. From *a* and *b* above, it is evident that the carrier source in conjunction with the rectifying units alternately places an open circuit or a short circuit across the output transformer.

81. Operating Principle of Balanced-Bridge Modulator

a. A, figure 106, shows a balanced-bridge modulator with a voice-voltage signal source and a carrier source. The instantaneous carrier-source polarity is such that G is negative and point E is positive; therefore, the rectifying units momentarily are open-circuited. Suppose the instantaneous polarity of the voice-voltage source is such that voice current flows down through R1 and, through transformer action, up through R2, as indicated by the arrows. No voice current goes through the rectifying units because they are open-circuited. Usually, resistance R1 and resistance R2 are equal; that is, the input and output circuits are matched. At some other instant of time, the carrier source will have the same polarity as shown in the figure, but the voice-voltage polarity will be reversed. When this occurs, the rectifying units still will be open-circuited, but the voice current will flow up through R1 and down through R2. Therefore, there will be a reversal in the direction of the voice current in R2.

b. A reversal of the *carrier*-source polarity results, however, in no flow of voice current in output resistance R2. B, figure 106, shows this reversal of carrier-source polarity. Now, G is positive and E is negative; therefore, the rectifying units are momentarily short-circuited. Thus, there is a short circuit across the output transformer; no voice current flows through it, and no voice current

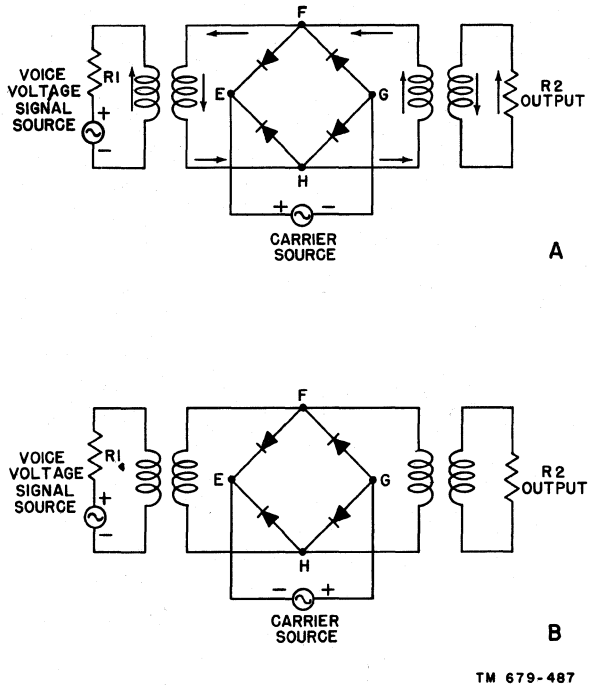


Figure 106. Operating principle of balanced-bridge modulator.

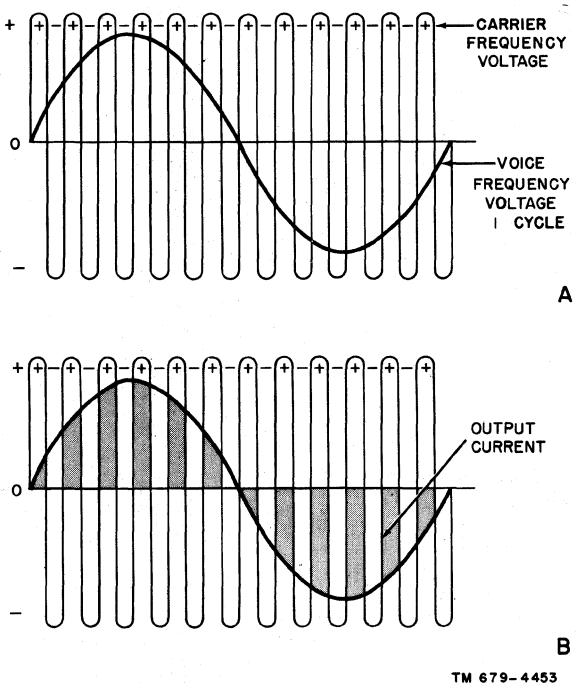


Figure 107. Development of output waveform of balanced-bridge modulator.

flows through resistance R2. This will occur regardless of the polarity of the voice source.

c. Assume that the carrier-source polarity that permits voice currents to flow through R2 is positive, and the reverse carrier-source polarity is negative. A, figure 107, shows one voice cycle of voltage and many carrier cycles of voltage on the same time axis, but not combined in the modulator. The positive and negative signs indicate the carrier-source polarity. For clarity, the carrier peak value is shown a little greater than that of the voice; actually, it is much greater. Voice current will flow through R2 during the positive carrier pulses only. The amount of voice current flowing through R2 during any positive carrier pulse depends on the instantaneous value of voice voltage, since the resistance of the varistor depends on the total voltage (carrier and voice) across it.

d. In B, figure 107, the shaded area represents the amount and polarity of voice *current* in modulator output R2 (fig. 106). The signal waveform in the modulator output is shown in figure 108. This waveform is made up of many different frequencies, but only a few are of any importance—the original voice frequency, the upper sideband, and the lower sideband. Although the original voice frequency is in the output, the original carrier frequency is not.

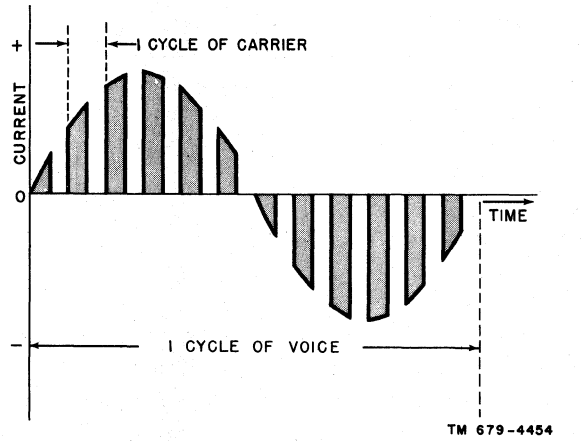


Figure 108. Output waveform of balanced-bridge modulator.

e. Figure 109, a simplified block diagram of a balanced-bridge modulator followed by a band-pass filter, emphasizes the location of several frequencies in the equipment at the transmitting terminal of a simple transmission system. The voice, V (1,000 cycles), and carrier, C (12,000 cycles), frequencies are combined in the modulator to produce the output waveform. The output waveform consists of V, (C—V), (C+V), and other frequencies. In terms of the numerical values given, the modulator output contains frequencies of 1,000, 11,000, and 13,000 cycles.

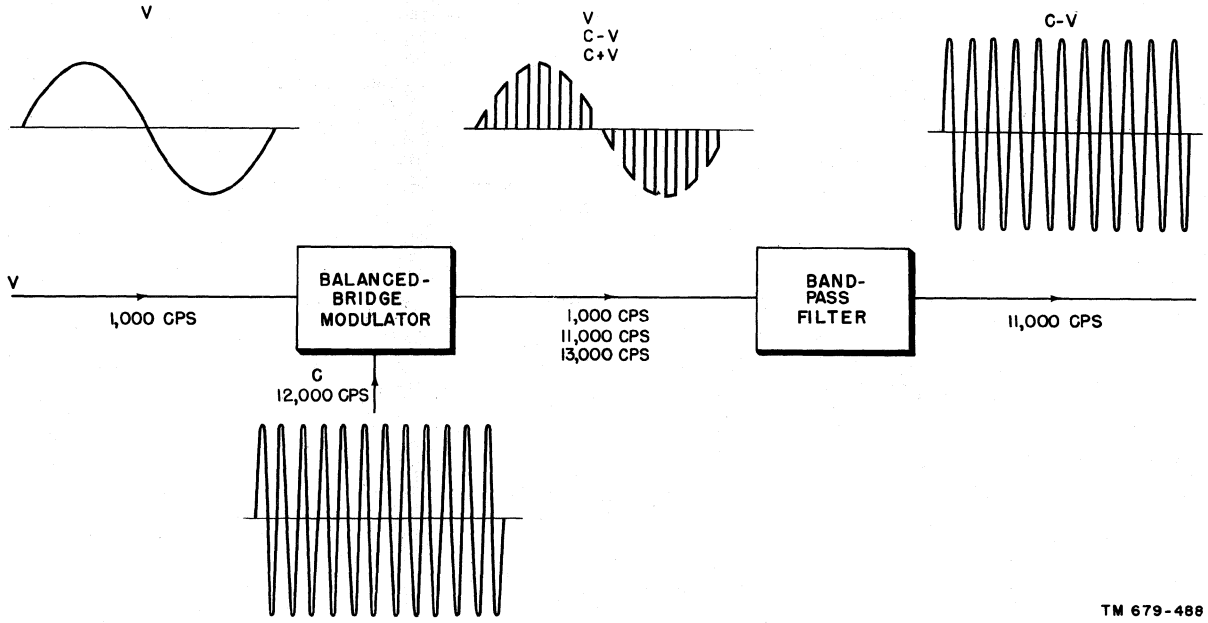


Figure 109. Location of frequencies at transmitting terminal.

The band-pass filter allows only one of the sidebands to pass, however—the modulator-output component of 11,000 cycles, which is the lower sideband (C—V).

82. Lattice-Type Modulator

Another circuit arrangement of varistor units or germanium-crystal diodes, used to obtain desired modulation or demodulation results in some carrier-telephone systems, is called the *lattice-type modulator*. Figure 110 shows this circuit arrangement in two forms. The individual rectifying units are numbered, to help show the equality of the forms.

a. A, figure 110, emphasizes the bridge arrangement of the rectifying units which is similar to that of the balanced-bridge modulator. Beyond this, the similarity ceases, for the manner in which the voice source, carrier source, and output are connected is different.

b. In figure 110, B is the conventional method of illustrating the lattice-type modulator. The voice-frequency voltage is applied by means of transformer coupling; the carrier-frequency voltage is applied at the center taps of the windings across opposite ends of the bridge. The output is across the secondary winding at the extreme right. The rectifying units used in a lattice-type modulator operate as carrier-controlled switches; therefore, the carrier voltage must be greater than the voice voltage. The units perform in a manner that reverses the direction of the voice-frequency current in the output at the rate of the carrier frequency. This lattice-type modulator combines the voice and carrier frequencies in such a way that the output contains only upper and lower sidebands; this is an advantage over the balanced-bridge modulator.

83. Carrier-Frequency Voltage-Switching in Lattice-Type Modulator

To understand the operation of the lattice-type modulator, it is necessary—as in the case of the balanced-bridge modulator—to know the exact manner in which the carrier-controlled switches are operated by the carrier voltage.

a. A, figure 111, shows a lattice-type modulator to which only carrier voltage is applied. R1 is the input impedance of the voice source and R2 is the impedance of the modulator load. The carrier source has the polarity indicated. This carrier-source polarity causes rectifier units 1 and 3 to be short-circuited and units 2 and 4 to be open-circuited. In B, the short-circuited condition of the series units and the open-circuited condition of the shunt units are emphasized. The path of the carrier current is indicated by arrows. At F, the current divides equally, and flows in opposite directions through the center-tapped winding; at E, the currents are combined

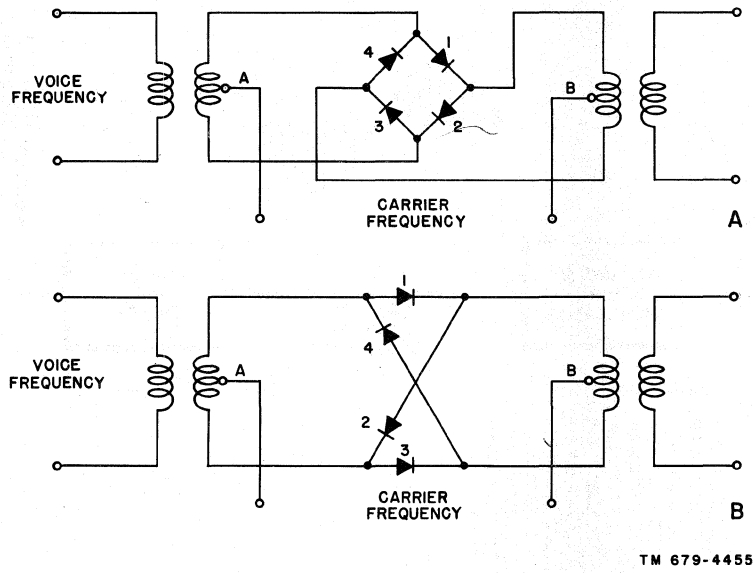


Figure 110. Lattice-type modulator.

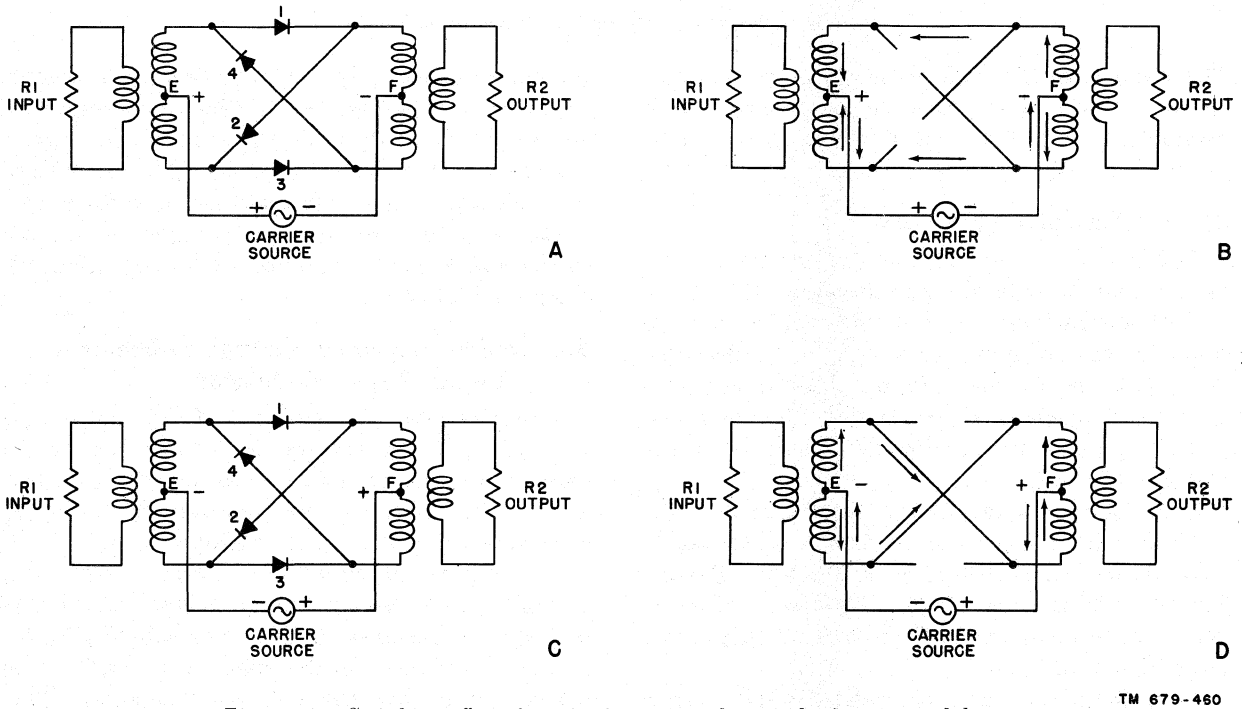


Figure 111. Switching effect of carrier frequency voltage in lattice-type modulator.

and flow back to the carrier source. Since equal currents flow in opposite directions through the center-tapped windings, no voltages are induced into the other windings, and no carrier current flows in resistances R1 and R2.

b. In C, figure 111, the carrier-source polarity is reversed, causing rectifier units 2 and 4 to be short-circuited and units 1 and 3 to be open-circuited. D emphasizes the short-circuited condition of the shunt units and the open-circuited condition of the series units. At F, the current divides equally, and flows in opposite directions through the center-tapped winding; at point E, the currents are combined, and flow back to the carrier source. Since equal currents flow in opposite directions through the center-tapped

windings, no voltages are induced into the other windings, and no carrier current flows in resistances R1 and R2.

c. Comparing E and F, it will be seen that the reversal of the carrier polarity changes (like a switch) the closed-circuit path between the two center-tapped windings. The output polarities are reversed with respect to the voice-input polarities at the carrier-frequency rate.

84. Operating Principle of Lattice-Type Modulator

a. A, figure 112, shows a lattice-type modulator with a voice source and a carrier source. The instantaneous carrier-source polarity is such that the series units are short-circuited and the shunt units are open-circuited. Simultaneously, the voice source induces voltage into the secondary of the input transformer, causing circulating voice currents. Note the polarity of the voice source; it determines the direction of voice-current flow in the output resistance R2. R1 is the impedance of the voice source, B shows the resultant circuit for the instantaneous voltage polarities illustrated in A. The direction of voice-current flow is indicated by arrows. Voice currents are induced into the secondary of the input transformer, which is center-tapped at point E. Equal currents are induced into each half-winding and flow in the same direction. This current circulates in the center loop and, in turn, causes induced current to flow upward in R2. The polarity of the carrier source will change before that of the voice source, however, resulting in a reversal of the direction of current in R2. Since equal voltages appear across the four half-windings, points E and F are at equal potential, as in an a-c Wheatstone bridge. Therefore no voice potential is applied across the carrier source and voice current does not flow through it.

b. In C, figure 112, the instantaneous polarity of the carrier source is reversed, causing the shunt units to be short-circuited and the series to be open-circuited. The direction of voice-current flow in the windings of the input transformer remains unchanged, as is indicated by arrows in D. The reversal of the carrier polarity changes the closed-circuit path between the two center-tapped windings; therefore, the current flows downward in resistance R2.

c. In this way, the carrier-source polarity controls the direction of output current for a given voice-source polarity. The amount of output current, however, depends at any instant upon the instantaneous value of voltage from the voice source. This means that as the direction of the

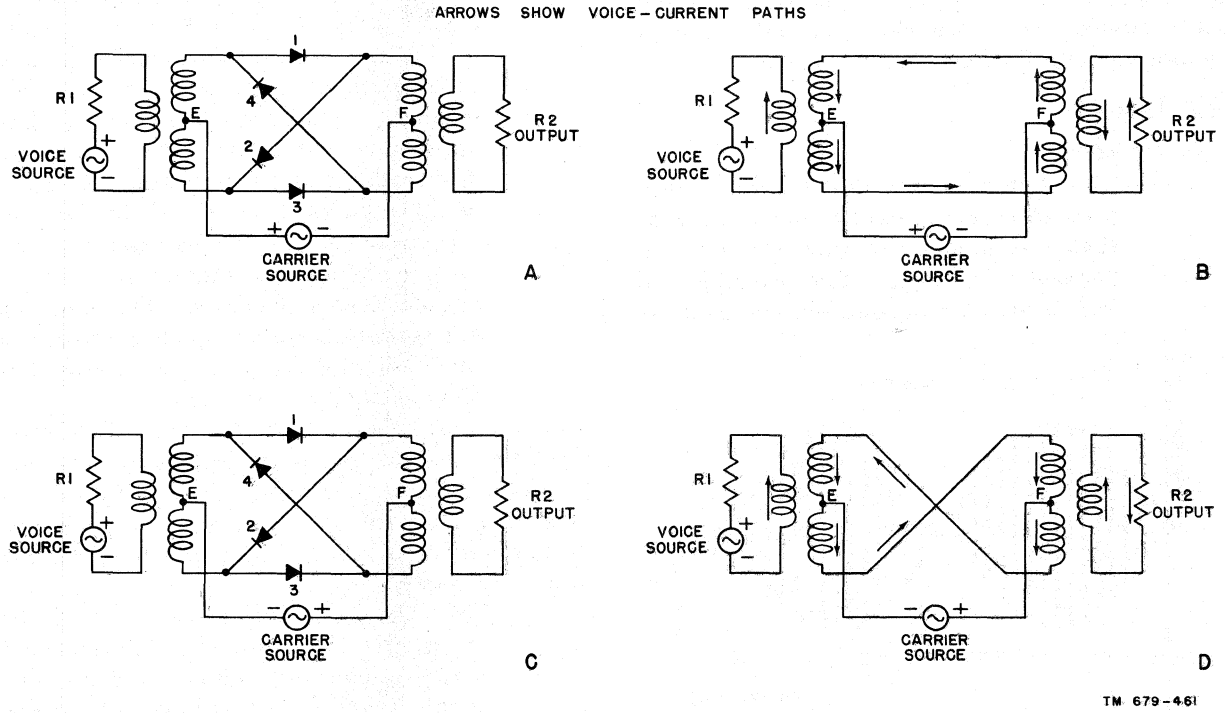


Figure 112. Operating principle of lattice-type modulator.

282463 O—54——7

output current changes at the carrier-frequency rate, its value changes in accordance with the value of voice-source voltage.

d. A, figure 107, shows one voice cycle and many carrier cycles (par. 81*c*). Assume that these are applied to the lattice-type modulator. Starting at zero time, the voice signal is increasing from zero, and the carrier signal is positive; this results in an increasing output current. During the next half-cycle of the carrier, direction of the output current is changed, but its value is increasing. This process continues, the envelope of the output current following the curve of the voice signal.

e. Figure 113 is the resultant waveform of the output current of the lattice-type modulator. Note the sharpness of change in current direction which is carrier controlled, and note that the current envelope is the voice-signal curve and that the positive and negative half-cycles are mirror images of each other. Current is always flowing in the output of the lattice-type modulator which is unlike the condition in the balanced-bridge modulator; therefore, there is more power in the output of the lattice-type modulator, and this, of course, is an advantage. Another advantage is the fact that its output contains only all of the upper and lower side bands. The carrier and voice signals are suppressed, but not completely, since the varistors do not present perfect open and short circuits and since the four half-windings cannot be perfectly balanced in practice.

85. Filters

a. Purpose.

(1) It is often necessary to eliminate or suppress frequencies that are unnecessary, or that will hinder the transmission of a particular message. For example, in the output of a balanced-bridge modulator there are many frequency components, but transmission of only one component is desired. The frequencies that are eliminated or suppressed are said to be *attenuated*. The circuit that is used to give a higher loss to some frequencies than it does to others is called an *electrical filter*, or, more commonly, a *filter*.

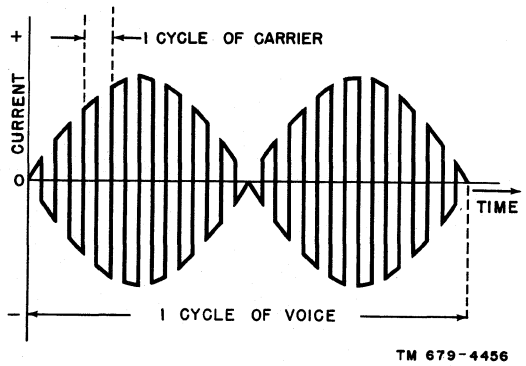


Figure 113. Output waveform of lattice-type modulator.

(2) Since filter loss varies with frequency, filters must consist of elements which have impedances that vary with frequency. Combinations of inductive and capacitive reactances fulfill this requirement. The formula for inductive reactance, $X_L=2\pi fL$, shows that as the frequency increases the inductive reactance increases also, which causes a greater opposition to the flow of alternating current. The formula for capacitive reactance, $X_C=\frac{1}{2}\pi fC$, shows that as the frequency increases the capacitive reactance decreases, which causes a smaller opposition to flow of alternating current.

(3) There are four types of filters—low-pass, high-pass, band-pass, and band-stop filters. The *low-pass filter* readily permits the passage of frequencies from zero up to a predetermined frequency, but attenuates all others. The *high-pass filter* readily permits the passage of frequencies from a predetermined frequency (not zero frequencies) up to infinite frequency, but attenuates all others. The *band-pass filter* readily permits the passage of a band of frequencies that lies between two predetermined frequencies (neither being zero frequency nor infinite frequency), but attenuates all others. The *band-stop filter* attenuates all frequencies within a specified band, but passes all frequencies above or below that band.

b. Principle. To perform filtering effectively, the filter must consist of at least two elements. In proof of this requirement, let us see how in a given load, an amount of current of high frequency is reduced, relative to an amount of current of low frequency.

(1) First, the use of only one element will be examined. A, figure 114, shows a load across a constant-voltage source. The source supplies two frequencies—a low frequency of 100 cycles and a high fre-

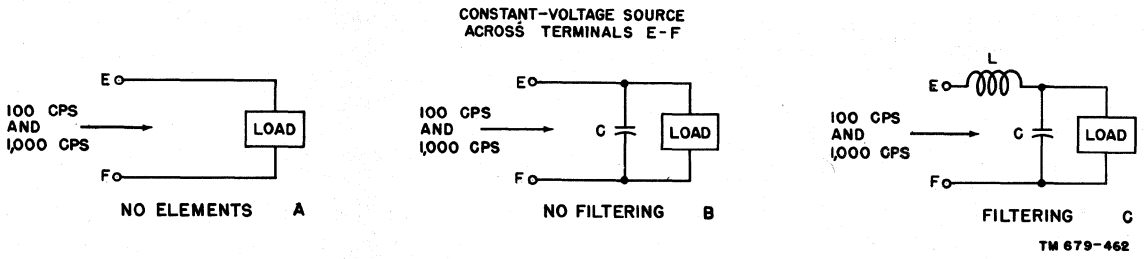


Figure 114. Principle of filter.

quency of 1,000 cycles. The reactance of a capacitor decreases as frequency increases; therefore place a capacitor, C, across the load, as in B; this in an attempt to reduce the amount of 1,000-cycle current compared to the 100-cycle current. The attempt fails. No reduction of either 100-cycle or 1,000-cycle current is obtained in the load, because the constant-voltage source is still directly across the load, and the load current is directly proportional to the voltage across the load. Therefore, a single element in shunt does not perform filtering.

(2) Now, two elements will be used. In order to reduce the relative amount of 1,000-cycle current in the load, it is necessary to reduce the 1,000-cycle voltage more than the 100-cycle voltage across the load. This is accomplished, in C, by the further addition of a series inductance, L, between the constant-voltage source and the shunt capacitance. The impedance that the constant-voltage source sees is the inductance in series with the parallel combination of the load capacitance. Recall that the voltages across the elements of a series circuit are directly proportional to the value of element impedance; also, that the inductive reactance is much greater at 1,000 cycles than at 100 cycles, and that the inverse is true for the capacitance reactance. Thus, the ratio of the inductive reactance to the total impedance of the parallel combination is much greater at 1,000 cycles than at 100 cycles, and the relative amount of 1,000-cycle voltage across the load is less than that for 100 cycles. (It can be much less.) This results in a relatively smaller amount of 1,000-cycle current in the load.

(3) If, in C, the shunt capacitance were omitted, the series inductance alone would perform filtering, but not as effectively. To perform filtering effectively, then, the filter must consist of at least two elements.

c. Filter. The precise values of the elements of a filter required to provide satisfactory filtering action in the frequency range desired may be computed mathematically. Examples of simple filter design are given in paragraphs 86 through 88. However, these are extremely simple cases, presented to illustrate the basic principles involved. In practice, the filter networks required for a given application are usually far more complicated, and their design often requires a great deal of careful calculation.

86. Low-Pass Filter

a. Arrangement of Low-Pass Filter. The arrangement of the inductance and capacitance shown in C, figure 114, attenuates the high frequencies, and this might therefore be considered a low-pass filter; but it is not a symmetrical network, such as usually is required for insertion between matched networks. To obtain a symmetrical network, another series inductance is added. It is placed between the shunt capacitance and the load, as shown in figure 115, where both the load and the voltage source have the same value of resistance, R. This arrangement is a *low-pass filter.*

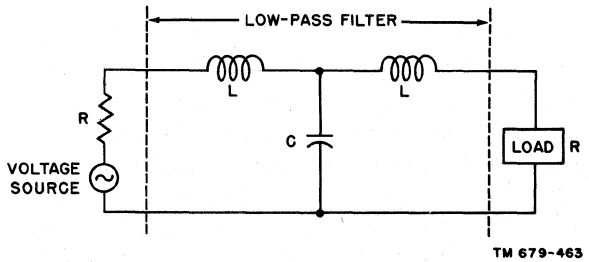


Figure 115. Low-pass filter.

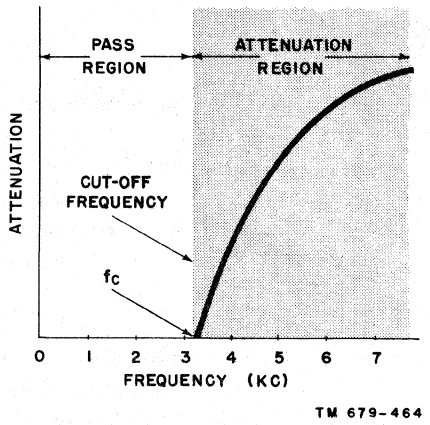


Figure 116. Frequency characteristic of low-pass filter.

b. Frequency Characteristic of Low-Pass Filter. Figure 116 shows the frequency characteristic of a low-pass filter. Attenuation (loss) is plotted vertically, and frequency is plotted horizontally. The *pass* region—from zero frequency to cutoff frequency, f_c—is the range of frequencies for which attenuation is zero. (Strictly, the attenuation in the pass region is not zero, but a small, negligible amount.) The cutoff frequency, f_c, is the frequency at which the attenuation starts to increase rapidly. The *attenuation region*—from the cutoff frequency to infinite frequency—is the range of frequencies for which the attenuation is appreciable.

c. Design of Low-Pass Filter. To design a low-pass filter, it is necessary to know the desired value of the cutoff frequency and the value of the load resistance to be matched. The design equations are:

(A) $$L=\frac{R}{2\pi f_c} \text{ henry},$$

(B) $$C=\frac{1}{\pi f_c R} \text{ farad},$$

where L is the series inductance, C is the shunt capacitance, and R is the load or source resistance, as in figure 115, and where f_c is the cutoff frequency, as in figure 116. For example, it is desired to design a low-pass filter for a 600-ohm circuit and a 3,000-cycle cutoff frequency. To obtain L and C, substitute the given values for R and f_c in equations (A) and (B) as follows:

(C) $$L=\frac{600}{2\pi(3{,}000)}=.0318 \text{ henry}=31.8 \text{ millihenrys}.$$

(D) $$C=\frac{1}{\pi(3{,}000)(600)}$$

$$=.177\times10^{-6} \text{ farad}=.177 \text{ microfarad}.$$

Therefore, in figure 115, each series inductance is 31.8 millihenrys, and the shunt capacitance is .177 microfarad.

87. High-Pass Filter

a. Principle of High-Pass Filter. Assume it is desired to suppress or eliminate the low-frequency 100-cycle current in the load of A, figure 114. Recall that, to perform filtering effectively, the filter must consist of at least two elements. One element is to be directly across the load and the other between the source and the load. The elements must be of such a nature as to reduce the 100-cycle voltage more than the 1,000-cycle voltage across the load. This is accomplished by the addition of a series capacitance and a shunt inductance, as in A, figure 117. The impedance that the constant-voltage source sees is the capacitance in series with the parallel combination of the load and inductance. Recall that the voltage across the elements of a series circuit is directly proportional to the value of element impedance; also, that the capacitive reactance is much greater at 100 cycles than at 1,000 cycles, and that the inverse is true for the inductive reactance. Thus, the ratio of the capacitive reactance to the total impedance of the parallel combination is much greater at 100 cycles than at 1,000 cycles, and the relative amount of 100-cycle voltage across the load is much less than that for 1,000 cycles. This results in a relatively smaller amount of 100-cycle current in the load.

b. Arrangement of High-Pass Filter. The arrangement of the capacitance and inductance in A, figure 117, is not a symmetrical network. Because filters usually are inserted between matched networks, another series capacitance is added to obtain a symmetrical network. It is placed between the shunt inductance and the load, as in B, where both the load and the voltage source have the same value of resistance R. This arrangement is a *high-pass filter.*

c. Frequency Characteristic of High-Pass Filter. Figure 118 shows the frequency characteristic of a high-pass filter. Attenuation is plotted vertically; frequency is plotted horizontally. The attenua-

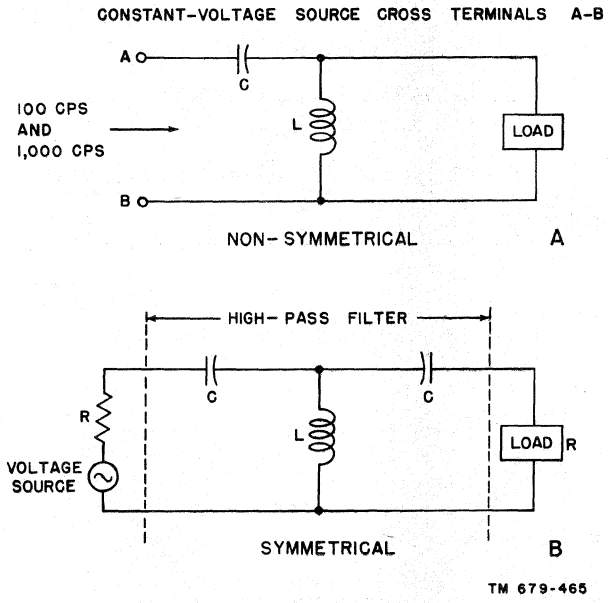


Figure 117. High-pass filter.

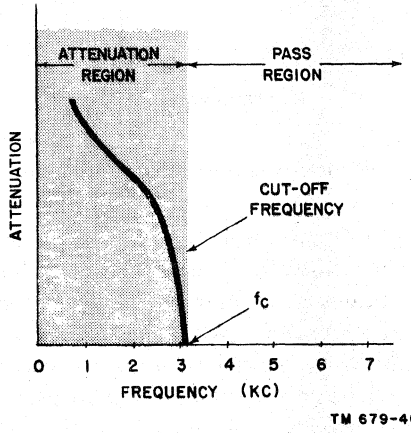


Figure 118. Frequency characteristic of high-pass filter.

tion region is from zero frequency to cut-off frequency, f_c; the pass region is from cutoff frequency, f_c, to infinite frequency. (Strictly, the attenuation in the pass region is not zero, but a small, negligible amount.)

d. Design of High-Pass Filter. To design a high-pass filter, it is necessary to know the desired valued of cutoff frequency and the value of the load resistance to be matched. The design equations are:

(A) $$L=\frac{R}{4\pi f_c} \text{ henry},$$

(B) $$C=\frac{1}{2\pi f_c R} \text{ farad},$$

where C is the series capacitance, L is the shunt inductance, and R is the load or source resistance, as in B, figure 117, and where f_c is the cutoff frequency, as in figure 118. For example, it is desired to design a high-pass filter for a 600-ohm circuit and a 5,000-cycle cutoff frequency. To obtain L and C, substitute the given values for R and f_c in equations (A) and (B) as follows:

(C) $$L=\frac{600}{4\pi(5{,}000)}=0.00955 \text{ henry}$$
$$=9.55 \text{ millihenrys},$$

(D) $$C=\frac{1}{2\pi(5{,}000)(600)}=0.053\times10^{-6} \text{ farad}$$
$$=0.053 \text{ microfarad}.$$

Therefore, in B, figure 117, each series capacitance is 0.053 microfarad, and the shunt inductance is 9.55 millihenrys.

88. Band-Pass Filter

To obtain a pass region between two frequencies, neither of which is zero or infinite, use is made of a low-pass filter in *cascade* (series) with a high-pass filter.

a. Pass-Band. Figure 119 shows the desired *pass-band* (pass region) of a *band-pass filter*. Attenuation is plotted vertically, frequency is plotted horizontally. There are two attenuation regions, separated by the pass-band. One attenuation region ranges from zero frequency to cutoff frequency f_1; the pass-band lies between cutoff frequencies f_1 and f_2; and the other attenuation region ranges from cutoff frequency f_2 to infinite frequency. To obtain this result, a low-pass *and* a high-pass filter, which have overlapping cutoff

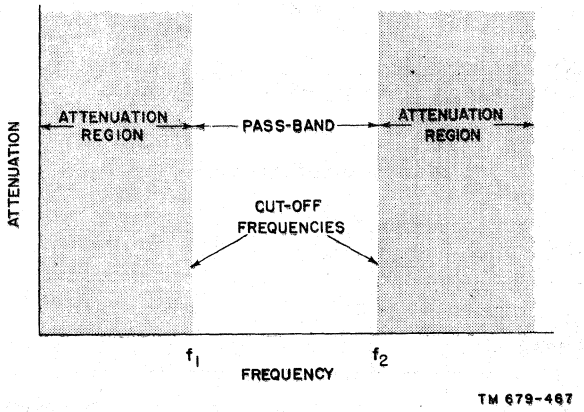


Figure 119. Analysis of pass-band.

frequencies, are arranged in cascade; that is, the cutoff frequency of the low-pass filter is greater than that of the high-pass filter. Only the range of frequencies between the two cutoff frequencies is in the pass region of both filters. All other frequencies are in the attenuation regions of either the low-pass or the high-pass filter.

b. Arrangement of Band-Pass Filter. Figure 120 shows one way in which a low-pass and a high-pass filter may be arranged in cascade and inserted between two matched networks to make a workable band-pass filter. Figure 121 shows an arrangement more commonly used. Each series arm consists of an inductance and a capacitance in series; the shunt arm is a parallel combination. This is a symmetrical network and is inserted between matched impedances—load impedance and source impedance.

c. Frequency Characteristic of Band-Pass Filter. The frequency characteristic of the filter in figure 121 is shown in figure 122. The pass-band is between cutoff frequencies f_1 and f_2. (As with the low- and high-pass filters individually, the attenuation in the pass region is not zero, but a small, negligible amount.)

d. Design of Band-Pass Filler. To design a band-pass filter, it is necessary to know the desired value of each cutoff frequency and the value of the load resistance to be matched. The design equations are:

(A) $$L1=\frac{R}{2\pi(f_2-f_1)} \text{ henry,}$$

(B) $$L2=\frac{(f_2-f_1)R}{4\pi f_1 f_2} \text{ henry,}$$

(C) $$C1=\frac{f_2-f_1}{2\pi f_1 f_2 R} \text{ farad,}$$

(D) $$C2=\frac{1}{\pi(f_2-f_1)R} \text{ farad,}$$

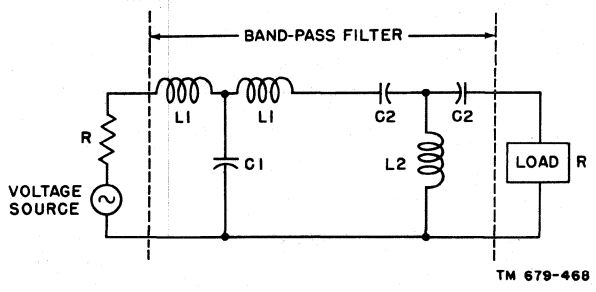


Figure 120. Workable band-pass filter.

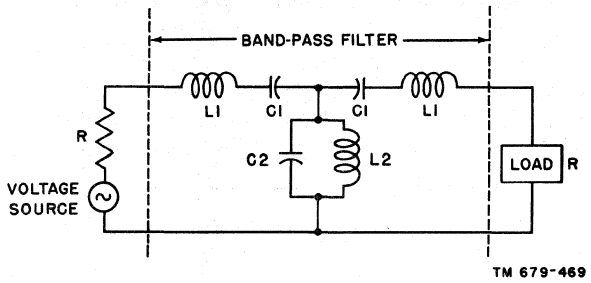


Figure 121. Commonly used band-pass filter.

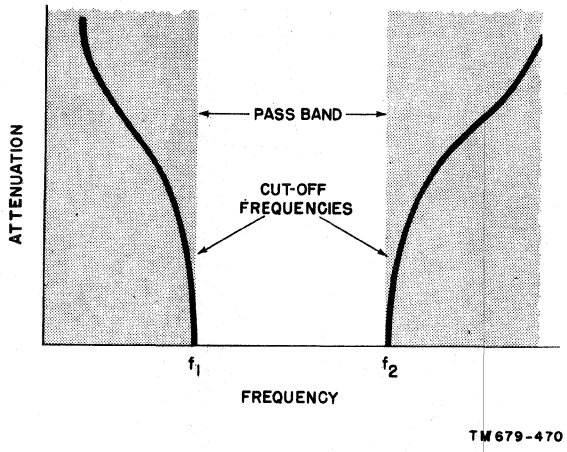


Figure 122. Frequency characteristic of band-pass filter.

where $L1$ and $C1$ comprise each series, $L2$ and $C2$ comprise the shunt arm, and R is the load or source resistance as in figure 121, and where f_1 and f_2 are the cutoff frequencies as in figure 122. For example, it is desired to design a band-pass filter for a 600-ohm circuit and a pass-band from 3,000 to 6,000 cycles. To obtain L1, C1, L2, and C2, substitute the given values for R, f_1, and f_2 in equations (A), (B), (C), and (D) as follows:

(A) $$L1=\frac{600}{2\pi(6{,}000-3{,}000)}$$
$$=0.0318 \text{ henry}=31.8 \text{ millihenrys,}$$

(B) $$L2=\frac{(6{,}000-3{,}000)(600)}{4\pi(6{,}000)(3{,}000)}$$
$$=0.00796 \text{ henry}=7.96 \text{ millihenrys,}$$

(C) $$C1=\frac{6{,}000-3{,}000}{2\pi(6{,}000)(3{,}000)(600)}$$
$$=0.0442\times10^{-6} \text{ farad}=0.0442 \text{ microfarad,}$$

(D) $$C2=\frac{1}{\pi(6{,}000-3{,}000)(600)}$$
$$=0.177\times10^{-6} \text{ farad}=0.177 \text{ microfarad.}$$

The values above, in the circuit of figure 121, gives a pass-band of 3,000 to 6,000 cycles.

e. Design of Band-Stop Filter. To design a band-stop filter, it is necessary to know the desired value of each cutoff frequency, and the value of the load resistance to be matched. The design equations for a pi-type band-stop filter are:

(A) $$L1=\frac{(f_2-f_1)R}{\pi f_1 f_2} \text{ henry,}$$

(B) $$L2=\frac{R}{\pi 2(f_2-f_1)} \text{ henry,}$$

(C) $$C1=\frac{1}{\pi 4(f_2-f_1)R} \text{ farad,}$$

(D) $$C2=\frac{(f_2-f_1)}{\pi 2 f_1 f_2 R} \text{ farad,}$$

where C is in farads, L is in henrys, R is in ohms, and f is in cycles per second. $L1$ and $C1$ comprise the series arm, $L2$ and $C2$ comprise the shunt arm, and R is the load or source resistance, as in figure 123. The cut-off frequencies are f_1 and f_2, as in figure 124.

For example, it is desired to design a band-stop filter for a 600-ohm circuit and a stop band from 3,000 to 6,000 cycles. To obtain L1, C1, L2, and C2, substitute the given values for R, f_1, and f_2 in equations (A), (B), (C), and (D) as follows:

(A) $$L1=\frac{(6{,}000-3{,}000)600}{\pi(3{,}000)(6{,}000)}=0.0318 \text{ henry}$$
$$=31.8 \text{ millihenrys,}$$

(B) $$L2=\frac{600}{2\pi(6{,}000-3{,}000)}=0.0318 \text{ henry}$$
$$=31.8 \text{ millihenrys,}$$

(C) $$C1=\frac{1}{4\pi(6{,}000-3{,}000)(600)}$$
$$=0.044 \text{ microfarad,}$$

(D) $$C2=\frac{(6{,}000-3{,}000)}{2\pi(3{,}000)(6{,}000)(600)}$$
$$=0.044 \text{ microfarad.}$$

The values above are for a pi-type band-stop filter as shown in figure 123, which has a band-stop characteristic from 3,000 to 6,000 cycles.

89. Pads

In telephone carrier sytems, unnecessary or increased attenuation of transmission *lines* is avoided so that maximum power is transmitted. In carrier *equipments*, however, it often is desirable

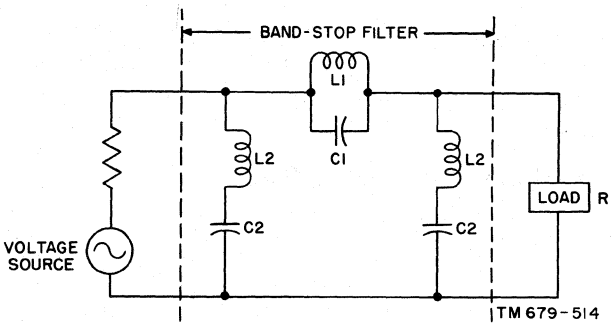


Figure 123. Commonly used band-stop filter.

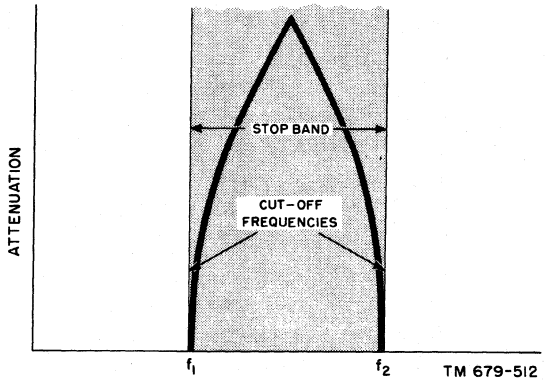


Figure 124. Frequency characteristic of band-stop.

to control the attenuation of signals and to attenuate their different frequency components equally. The *resistance pad*, or, simply, *pad*, is a network which does this. As the name implies, the pad consists only of resistive elements, which do not change their values with frequency changes.

a. Uses. In general, a symmetrical pad is inserted between matched networks for attenuation purposes only, and a nonsymmetrical pad is inserted between mismatched networks for attenuation purposes and to perform matching, as well. Only symmetrical pads are considered in this manual. Specific uses of pads are as follows:

(1) They are used to reduce the strength of voice signals and of incoming sidebands prior to modulation or demodulation.
(2) They are placed at the input of fixed gain amplifiers as a means of obtaining overall gain variations.
(3) They are placed at the input of amplifiers to prevent amplifier overloading.
(4) In conjunction with a voltmeter, they can be used to measure the gain of an amplifier.
(5) They are used to improve impedance matching in some cases.

b. Types. Four types of pads are the T, the pi, the H, and the square pad. The symmetrical versions of these four types are discussed in paragraphs 91 through 94.

90. Basic Attenuation Theory

a. Series-Resistance Method.

(1) One simple method of reducing the strength of the signal transmitted from one network to another is the series-resistance method shown in A, figure 125. The network from which the signal is being transmitted is represented by the a-c generator, E, and its internal resistance, R_g; the network receiving the signal is represented by resistance, R_L. The series resistance, R_s, is inserted between these two networks.

(2) R_s reduces the signal strength that reaches R_L(A, fig. 125). The voltages across the elements of a series circuit are directly proportional to the resistance of the elements, and the power delivered to a resistance is equal to the square of the voltage across the resistance divided by the resistance ($P=E^2/R$); therefore, with R_s in the circuit, the voltage across R_L is less than it would be without it, and, the voltage being less, the power (signal strength) delivered to R_L is less. Consequently, the insertion of a series resistance attentuates a transmitted signal. The *larger* the series resistance, the greater is the attentuation.

(3) This is not a satisfactory method of attentuation between two matched networks, however, for when R_g and R_L are equal, as in A, the insertion of R_s results in mismatched networks, causing unwanted reflection of signals and a harmful echo effect. Symmetrical pads are needed between matched networks.

b. Shunt-Resistance Method.

(1) Another simple method of reducing the strength of the signal transmitted from one network to another is the *shunt-resistance method* shown in B. This circuit is the same as that in A, except that a shunt resistance, R_p, is used instead of a series resistance.

(2) R_p reduces the signal strength that reaches R_L, in B. With R_p in the circuit, the total resistance seen by the generator is less, the current flow is greater, there is a larger voltage drop across R_g, and the voltage across R_L is less. Since a reduction in voltage across a resistance causes a reduction of the power delivered to it, less power is sent to R_L. Hence, the insertion of a shunt resistance attentuates a transmitted signal. The *smaller* the shunt resistance, the greater is the attenuation.

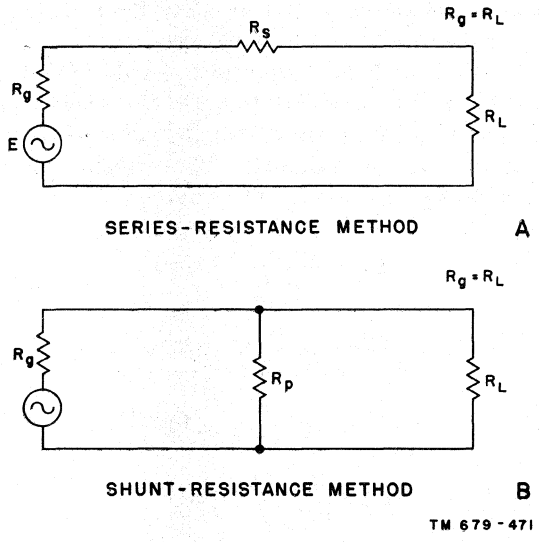


Figure 125. Attenuation by simple series and shunt methods.

(3) This, however, is not a satisfactory method of attentuation between two matched networks for, as with the series-resistance method, the result is mismatched networks, with unwanted reflection of signals and a harmful echo effect. Symmetrical pads are needed to provide attentuation between matched networks, and still not destroy the impedance match. To obtain them, series and shunt resistances are used in combination.

91. Symmetrical T pad

a. Description and Uses. A *symmetrical T pad* is represented in figure 126. It is inserted between two matched networks, since R_g and R_L are equal. It is symmetrical because it consists of two *equal* series resistances $R1$, one on each side of the junction with shunt resistance $R2$. This pad is not balanced to ground and consequently its use is restricted. It is employed when one side (the side not containing series resistances) of the attached networks is grounded, or when it is not

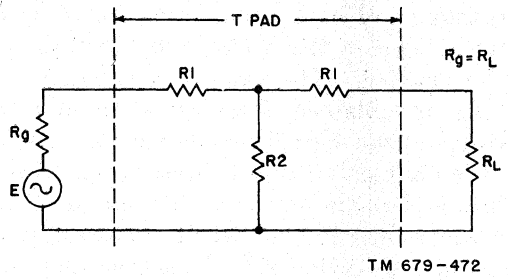


Figure 126. Symmetrical T pad.

necessary to have the attached networks balanced to ground.

b. Design. In order to design a symmetrical T pad (fig. 126) it is necessary to know the desired attenuation and the resistance value of the circuit into which the pad is to be inserted. This pad can be designed either by direct application of the design equations or by means of a table derived from the design equations. The design equations for a symmetrical T pad are—

$$\text{(A)}\quad R1 = R\frac{k-1}{k+1}\ \text{ohms},$$

$$\text{(B)}\quad R2 = R\frac{2k}{k^2-1}\ \text{ohms},$$

$$\text{(C)}\quad \text{Pad loss (in } db) = 20\log k,$$

where $R1$ is the resistance of each series arm (fig. 126), $R2$ is the shunt resistance, and R is the resistance value of the circuit into which the pad is to be inserted. The value k in equations (A) and (B) is obtained from equation (C) and knowledge of the desired pad loss in *db*. For the T pad, the circuit-resistance value, R, is multiplied by the series-arm factor, $\frac{k-1}{k+1}$, in equation (A) to obtain the series-arm resistance, and R is multiplied by the shunt-arm factor, $\frac{2k}{k^2-1}$, in equation (B) to obtain the shunt-arm resistance. Assume that it is desired to obtain the series- and shunt-arm factors for a pad loss of 20 *db*. Because the multiplier of R in each case depends on the value of k, it is necessary first to obtain k. This is accomplished by employing equation (C). For a pad loss of 20 *db*,

$$\text{Pad loss (in } db) = 20\log k,$$

$$\text{or } 20 = 20\log k,\ \text{or } 1 = 1\log k.$$

Since the antilog of 1 is 10, $k=10$. For $k=10$, the series-arm factor is—

$$\frac{k-1}{k+1} = \frac{10-1}{10+1} = .8182.$$

For $k=10$, the shunt-arm factor is—

$$\frac{2k}{k^2-1} = \frac{2(10)}{(10)^2-1} = 0.2020.$$

If the resistance of the circuit R is 500 ohms, $R1 = 500\times 0.8182$, or 409.1 ohms, and $R2 = 500\times .2020$, or 101 ohms. The factors in table III are obtained by application of the preceding equations to losses of 1 to 48 db in unit steps. The columns give the series-arm factors and shunt-arm factors for successive values of pad loss in db. In designing a symmetrical T pad from a table, it is convenient to use design equations (A) and (B) in terms of the series- and shunt-arm factors. They become—

$$\text{(D)}\quad R1 = R \times \text{series-arm factor},$$
$$\text{(E)}\quad R2 = R \times \text{shunt-arm factor}.$$

Assume that it is desired to insert a symmetrical T pad having a 12-db loss into a 600-ohm network.

Table III. Shunt- and Series-Arm Factors for T Pads Between Circuits of Equal Impedance

db	Series-arm factor	Shunt-arm factor	db	Series-arm factor	Shunt-arm factor
1	0. 0575	8. 6709	25	0. 8935	0. 1128
2	. 1146	4. 3046	26	. 9046	. 1005
3	. 1710	2. 8384	27	. 9145	. 0901
4	. 2262	2. 0966	28	. 9234	. 0797
5	. 2802	1. 6448	29	. 9315	. 0711
6	. 3323	1. 3386	30	. 9387	. 0633
7	. 3825	1. 1160	31	. 9452	. 0564
8	. 4305	. 9462	32	. 9510	. 0503
9	. 4762	. 8118	33	. 9562	. 0448
10	. 5195	. 7028	34	. 9609	. 0399
11	. 5603	. 6123	35	. 9651	. 0356
12	. 5985	. 5362	36	. 9688	. 0317
13	. 6342	. 4714	37	. 9721	. 0283
14	. 6673	. 4156	38	. 9751	. 0252
15	. 6980	. 3673	39	. 9778	. 0224
16	. 7264	. 3251	40	. 9802	. 0200
17	. 7525	. 2883	41	. 9823	. 0178
18	. 7764	. 2558	42	. 9842	. 0159
19	. 7982	. 2273	43	. 9859	. 0142
20	. 8182	. 2020	44	. 9875	. 0126
21	. 8363	. 1797	45	. 9889	. 0112
22	. 8528	. 1599	46	. 9900	. 0100
23	. 8678	. 1423	47	. 9911	. 0089
24	. 8813	. 1267	48	. 9921	. 0080

From table III, series-arm factor for 12 db is 0.5985 and the shunt-arm factor is 0.5362. Substituting these values into equations (D) and (E),

$$R1=(0.5985)(600) \text{ ohms}=359.1 \text{ ohms},$$
$$R2=(0.5362)(600) \text{ ohms}=321.72 \text{ ohms}.$$

Therefore (fig. 126), each series resistance is 359.1 ohms and the shunt resistance is 321.72 ohms for a T pad that causes a 12-db loss in a 600-ohm circuit.

92. Symmetrical Pi Pad

a. Description and Uses. A *symmetrical pi pad* is represented in figure 127. It is inserted between two matched networks, R_g and R_l being equal. The pad consists of two equal shunt resistances $R2$ and series resistance $R1$ located between the shunt arms. This pad is not balanced to ground, and therefore its use is restricted. It is employed when one side (the side not containing the series resistance) of the attached networks is grounded, or when it is not necessary to have the attached networks balanced to ground.

b. Design. To design a symmetrical pi pad (fig. 127) it is necessary to know the desired pad attenuation and the resistance value of the circuit into which the pad is to be inserted. This pad can be designed by direct application of the design equations or by means of a table derived from the design equations.

The design equations for a symmetrical pi pad are—

$$\text{(A)}\quad R1=R\frac{k^2-1}{2k}=\text{ohms},$$

$$\text{(B)}\quad R2=R\frac{k+1}{k-1},$$

$$\text{(C)}\quad \text{Pad loss (in } db)=20 \log k,$$

where $R1$ is the series resistance (fig. 127), $R2$ is the resistance of each shunt arm, and R is the resistance value of the circuit into which the pad is to be inserted. The value k in equations (A) and (B) is obtained from equation (C) and knowledge of the desired pad loss in *db*. Assume that it is desired to insert a 20-*db* symmetrical pi pad into a 600-ohm network. To obtain $R1$ and $R2$ in equations (A) and (B), it is necessary first to determine the value of k. This is accomplished by employing equation (C). Since the pad loss is 20 *db*, pad loss (in *db*)$=20 \log k$, or $20=20 \log k$, or $1=\log k$. Since the antilog of 1 is 10, $k=10$. For $k=10$, and $R=600$ ohms, equations (A) and (B) yield

$$R1=600\frac{(10)^2-1}{2(10)} \text{ ohms, or 2,970 ohms},$$

$$R2=600\frac{10+1}{10-1} \text{ ohms, or 733.3 ohms}.$$

Therefore, the series resistance is 2,970 ohms and each shunt resistance is 733.3 ohms for a pi pad that causes a 20-*db* attenuation in a 600-ohm circuit (fig. 127).

A table for the design of symmetrical pi pads can be constructed in the same manner as table III for symmetrical T pads. For the pi pad, the circuit-resistance value, R, is multiplied by the series-arm factor, $\frac{k^2-1}{2k}$, in equation (A) to obtain the series-arm resistance, and R is multiplied by the shunt-arm factor, $\frac{k+1}{k-1}$, in equation (B) to obtain the shunt-arm resistance.

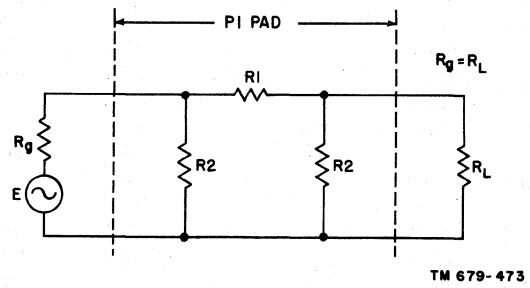


Figure 127. Symmetrical pi pad.

93. Symmetrical H Pad

a. Description and Uses. A *symmetrical H pad* is represented in figure 128. It is derived directly from the symmetrical T pad of figure 126. Each series resistance, $R1$, of the T pad is divided into two equal parts, one of which is placed in the opposite line on the same side of $R2$. This symmetrical H pad is balanced to ground, and therefore does not have the restrictions in use that the T pad has. It is employed when the attached networks are balanced to ground, never when one line side of the attached networks is grounded.

b. Design. The design equations of the H pad, like the pad itself, are derived directly from the T pad; therefore, this pad can be designed either by direct application of design equations or by

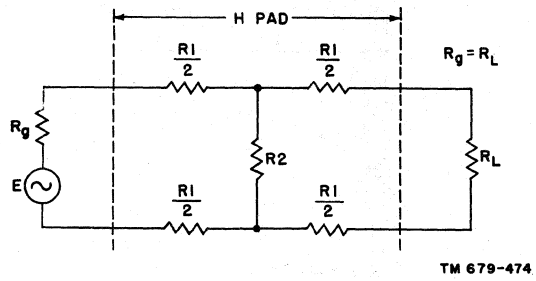


Figure 128. Symmetrical pi pad.

use of table III. The design equations for a symmetrical H pad are—

$$\text{(A)} \qquad \frac{R1}{2}=\frac{(R)\,(k-1)}{(2)\,(k+1)},$$

$$\text{(B)} \qquad R2=R\,\frac{2k}{k^2-1},$$

$$\text{(C)} \qquad \text{Pad loss (in db)}=20\log k,$$

where $R1/2$ is the value of each series resistance, $R2$ is the shunt resistance (fig. 128), and R is the resistance value of the circuit into which the H pad is to be inserted. The value k is obtained as it is for the T pad. Equations (A) and (B) may be written—

$$\text{(D)} \qquad \frac{R1}{2}=\frac{R}{2}\times\text{series-arm factor},$$

$$\text{(E)} \qquad R2=R\times\text{shunt-arm factor}.$$

The series- and shunt-arm factors are those listed in table III; therefore, use of this table reduces the amount of necessary calculation in the design of symmetrical H pads. Assume that it is desired to insert a 15-db pad into a 600-ohm network. Table III shows that for a loss of 15 db the series-arm factor is .6980 and the shunt-arm factor is 0.3673. Substituting these values into equations (D) and (E) yields

$$\frac{R1}{2}=\frac{(600)\,(0.69880)}{(2)}\text{ ohms, or 209.40 ohms,}$$

$$R2=(600)(0.3673)\text{ ohms, or 220.38 ohms.}$$

Therefore (fig. 128), each series resistance is 209.40 ohms and the shunt resistance is 220.38 ohms for an H pad that causes a 15-db loss in a 600-ohm circuit.

94. Symmetrical Square Pad

a. Description and Uses of Square Pad. A *symmetrical square pad* is represented in figure 129. It is derived directly from the symmetrical pi pad of figure 127. The series resistance $R1$ of the pi pad is divided into 2 equal parts, 1 of which is placed in the opposite line between the shunt resistances. This symmetrical square pad is balanced to ground, and therefore does not have the restrictions in use that the pi pad has. It is employed when the attached networks are balanced to ground, but never when one line side of the attached networks is grounded.

b. Design. The design equations of the square pad, like the pad itself, are derived directly from the pi pad; therefore, this pad can be designed either by direct application of design equations or by means of a table derived from the design equations. The design equations for a symmetrical square pad are—

$$\text{(A)} \qquad \frac{R1}{2}=\frac{(R)\,(k^2-1)}{(2)^2k}\text{ ohms,}$$

$$\text{(B)} \qquad R2=R\,\frac{k+1}{k-1}\text{ ohms,}$$

$$\text{(C)} \qquad \text{Pad loss (in db)}=20\log k,$$

where $R1/2$ is the value of each series resistance, $R2$ is a shunt resistance (fig. 129), and R is the resistance value of the circuit into which the pad is to be inserted. The value k is obtained as it is for the pi pad. Examination of design equations (A) and (B) reveal that they contain the pi-pad series- and shunt-arm factors. In (A), paragraph 92*b*, a 20-*db* symmetrical pi pad is designed for use in a 600-ohm circuit. This pad can be converted into a symmetrical square pad simply by dividing the series resistance, $R1$, of the pi pad into two equal parts. Therefore (fig. 129), each series

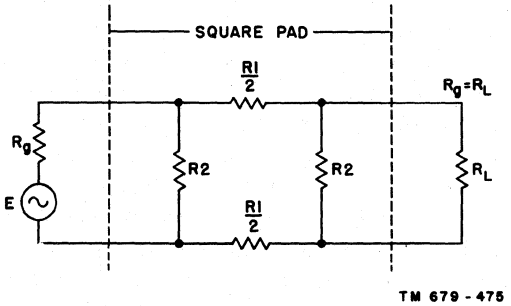


Figure 129. Symmetrical square pad.

resistance is 1,485 ohms and each shunt resistance is 733.3 ohms for a square pad that causes a 20-*db* loss in a 600-ohm circuit.

95. Non-Interchangeability of Pad-Design Equations

Although the design equations for the elements of the T and pi pads are of the same form, they cannot be interchanged, nor can the elements of one be rearranged directly to obtain the other. This is true for the H pad and the square pad also. In a circuit having the side of one line grounded, a T or a pi pad would be employed for attenuation purposes, whereas in a circuit balanced to ground an H or a square pad would be used.

96. Equalizers

a. Need for Equalizers. The attenuation of a transmission line is not the same at all frequencies. High frequencies are attenuated more than low frequencies, resulting in distortion. Since the attenuation for any frequency is directly proportional to the length of the line, the amount of distortion increases with increase in the length of the line. For this reason, it is frequently necessary to use a network to correct unequal attenuation in long transmission lines. Such a network is called an *attenuation equalizer*, or, simply, an *equalizer*. It adds attenuation, to make the total attenuation the same for all frequencies. Since the equalizer does not attenuate all frequencies equally, it must consist of inductors and capacitors, in addition to resistors.

b. Equalizer and Line Characteristics. Figure 130 is a plot illustrating the principle of the equalizer. It shows a typical line-loss characteristic and the desired equalizer-loss characteristic. Loss is plotted vertically, frequency horizontally. The line loss increases with an increase in frequency, and the equalizer loss is the inverse of the line loss; if the equalizer-loss curve were inverted, it would coincide with the line-loss curve. The total-loss curve of the line and equalizer is constructed by summing the line- and equalizer-loss characteristics. As shown, the desired total loss of line and equalizer is the same for all frequencies. Since the total loss is a constant value, the equalizer can be inserted into the circuit at either the sending or the receiving end. It is more commonly inserted at the receiving end.

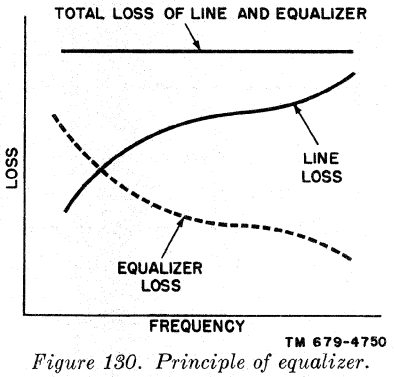


Figure 130. Principle of equalizer.

c. Operation of Equalizer.

(1) Equalizer networks, like filters, can be very complicated. Their principle of operation, however, can be understood from the simplified circuit of figure 131, which shows a simplified version of an equalizer placed before the amplifier at the receiving end of carrier system. (Placement at the input of the amplifier rather than at the output is preferable, for it prevents overloading of the amplifier by the frequencies least attenuated by the line.) The equalizer consists of a variable resistor, R, inductor, L, and capacitor, C. This equalizer has a loss characteristic similar to the equalizer-loss curve in figure 130; that is, its loss decreases with an increase of frequency. In other words, the shunting effect of the equalizer decreases with an increase in frequency.

(2) To bring this about, the impedance of the equalizer has to increase with an increase in frequency. The impedance of the series combination of R and L (fig. 131) alone would give this result, but the resultant impedance would not increase at the desired rate; therefore, capacitor C is required.

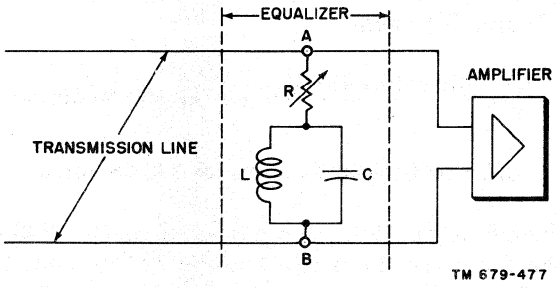


Figure 131. Equalizer.

(3) The addition of capacitor C (fig. 131) causes the impedance of the equalizer to increase at a greater rate with an increase in frequency, because the parallel-resonant frequency is above the transmission-frequency band, and with C in the circuit the impedance at the resonant frequency is much greater. Thus, the equalizer impedance must increase at a greater rate as frequency increases, to reach this very large impedance value.

(4) The amount of current in resistor R decreases with an increase in frequency (fig. 131). Since the power dissipated is equal to the product of the current squared and the resistance, the power dissipated in the equalizer decreases with an increase in frequency. The amount of dissipated power also can be varied by the variable resistor, R.

(5) Figure 132 is an illustration of the front view of a typical equalizer panel. There are four variable controls—low-frequency, flat-gain, high-frequency, and mid-frequency. These controls readily permit obtaining the correct equalizer loss characteristic for transmission lines of different lengths. The equalizer chart is used to record control settings for different line lengths; this permits rapid equalizer alinement.

d. Automatic Equalization. Although equalizers in most tactical equipments are usually the adjustable type, the recently designed carrier equipments feature automatic equalization. Pilot frequency regulation is used to provide a flat or a curved gain effective over the full band or part of the band.

97. Limiters

a. In carrier systems, momentarily large peak values of signal power might cause poor communication if networks called *limiters* were not employed. Limiters automatically reduce large peak values. A limiter is used in the voice-frequency channel, for example, to prevent overloading of the common transmitting amplifier.

b. The limiters in telephone carrier equipments widely use copper-oxide varistors or germanium-crystal diodes. Two types of limiters now to be discussed are the directly connected and the magnetically coupled types.

98. Directly Connected Limiter

a. Description.

(1) Figure 133 shows a *directly connected limiter*. The input to the circuit is across terminals A–B. There are two branches across the input—one, the limiter in series with transformer winding 8–9; the other, transformer winding 7–8. The transformer consists of three windings: 7–8, the input winding; 8–9, the limiter winding; and 10–11, the output winding. The output winding is across terminals C–D. The limiter consists of two parallel pairs of rectifying units shunted by equal resistors, M and N, and a small-voltage bias battery. The bias voltage causes points 1 and 4 to be negative with respect to points 2 and 5, and points 2 and 5 negative with respect to points 3 and 6.

(2) For this condition, the varistors exhibit a very high resistance (fig. 103). When the copper side of a copper-oxide varistor

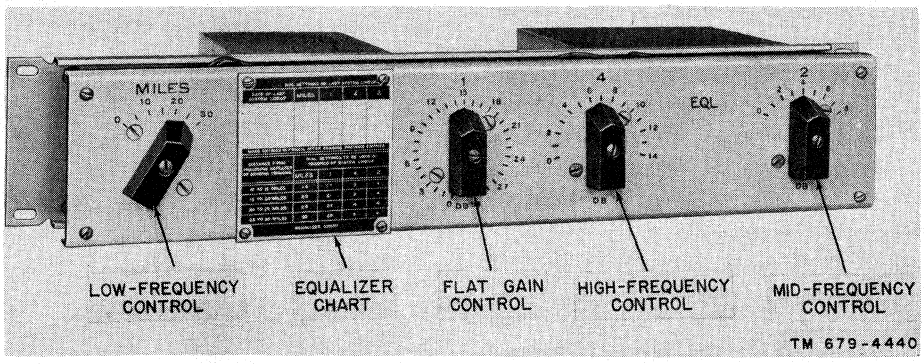


Figure 132. Equalizer panel, front view.

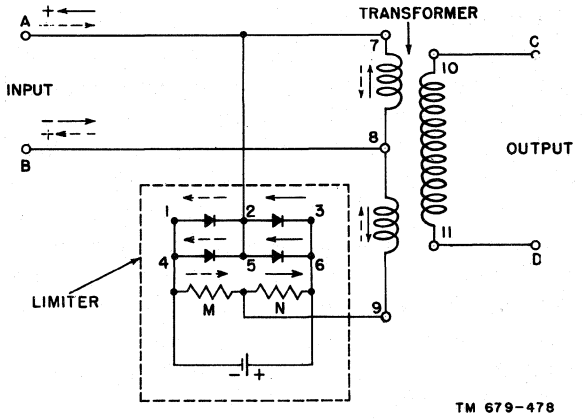


Figure 133. Directly connected limiter.

is only very slightly positive with respect to the copper-oxide side, the varistor has a large value of resistance, or is in an open-circuit condition; it is when the voltage is reversed and increased to about one-half volt or more that the varistor has a small value of resistance, or approaches a short-circuit condition. In the circuit as used in practice, the bias voltage is obtained from the vacuum-tube plate-voltage supply by way of a large dropping resistor.

b. Operation.

(1) Assume that there is a large peak (instantaneous) value of voltage across A–B (fig. 133), causing B to be negative and A positive as indicated by the solid plus and minus signs. This makes 1 and 4, and 3 and 6 negative with respect to 2 and 5. The rectifying units to the left of 2 and 5 remain open-circuited; but the input voltage makes 2 and 5 sufficiently positive with respect to 3 and 6 to short-circuit the pair of rectifying units to the right. Thus current flows from B (solid arrow) downward through winding 8–9, through resistor N, through these short-circuited varsitors, to terminal A. Current also flows upward in winding 7–8. The current flow in winding 8–9 is smaller than that in winding 7–8, because of the additional resistance, N, but is in the opposite direction, and consequently the voltage in secondary winding 10–11 is reduced. In this way a large peak value of voltage at the input is *limited* at the output.

(2) For an input voltage of reversed polarity, as indicated by the dashed minus and plus signs, the result is the same (fig. 133). Points 2 and 5 are made negative with respect to 1 and 4, and 3 and 6. The rectifying units between 1 and 4 and 2 and 5 are short-circuited, the other pair remaining open-circuited; current flows downward in winding 7–8 and upward in winding 8–9; thus, again, a large peak voltage at the input (A negative and B positive) is limited at the output. For normal values of input voltage, the rectifying units are open-circuited and no current flows in the limiter winding; thus, normal input-voltage values are not limited.

99. Magnetically Coupled Limiter

a. Description. Figure 134 shows a *magnetically coupled limiter.* The input to the circuit is across terminals A–B. Transformer winding 7–8, across the input, is magnetically coupled to limiter winding 9–10 and output winding 11–12. Limiter winding 9–10 is across the limiter, and output winding 11–12 is across terminals C–D. The limiter consists of two parallel pairs of rectifying units shunted by equal resistors, M and N, and a small-voltage bias battery. The bias voltage performs the same functions as in the directly connected limiter, and is obtained in the same way.

b. Operation.

(1) Assume there is a large peak (instantaneous) value of voltage across A–B (fig. 134), A being negative and B positive (solid minus and plus signs). Current flows downward in winding 7–8 and upward in winding 9–10 (solid arrows). The large input voltage induces a large

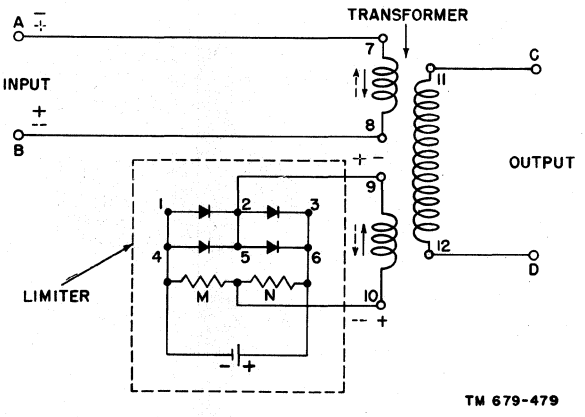


Figure 134. Magnetically coupled limiter.

voltage into limiter winding 9–10, making 9 negative and 10 positive. The rectifying units to the right of 2 and 5 remain open-circuited; but the large induced voltage makes 2 and 5 sufficiently negative with respect to 1 and 4 to short-circuit the pair of rectifying units to the left, thus making a path for current through these units and resistor M to 10 and upward through winding 9–10. The impedance reflected back into the primary winding 7–8 increases and causes the primary current to decrease. As a result, less voltage is induced in the secondary winding 11–12. Thus, a large peak value of voltage at the input is *limited* at the output.

(2) For an input voltage of reversed polarity (dotted plus and minus signs), the result is the same (fig. 134). The rectifying units between 3 and 6 and 2 and 5 are short-circuited, the other pair remaining open-circuited; current flows upward in 7–8 and downward in 9–10; thus, again, a large peak voltage at the input (A positive and B negative) is limited at the output. For normal values of input voltage, the rectifying units are open-circuited, and no current flows in the limiter winding; thus, normal input-voltage values are not limited.

100. Carrier Terminals

In practice, the components of carrier systems (hybrids, pads, modulators, and so on) are incorporated into *carrier terminals*, which are connected between the switchboard and the transmission medium. As a rule, the components constitute subassemblies on separate chassis, mounted on racks, the whole group inclosed in a suitable case or cabinet. There are two main kinds of carrier terminals—four-wire carrier terminals and two-wire carrier terminals.

101. Typical Four-Wire Carrier Termina!

Figure 135 is a simplified block diagram of a *typical four-wire carrier terminal.* It can be said to consist of three major groups of components—the voice channel, the carrier channel, and the common amplifiers. For simplicity, the power supply needed for the operation of the various electronic-tube circuits is not shown. Each channel is terminated in a two-wire circuit at one end and a four-wire circuit at the other end. The two-wire side of each channel goes to a voice-frequency ringer, thence to a switchboard. The four-wire side of each channel goes to a common amplifier, to the equalizer panel, and thence to the transmission line. Only two channels are shown, but other carrier channels can be added; they would be placed in parallel on the four-wire side prior to the common amplifiers. Such additional carrier channels require carrier frequencies so spaced out that the channel frequency bands do not interfere with each other.

102. Operation of Carrier Channel of Four-Wire Terminal

Figure 135 shows that each carrier channel consists of two branches, the transmitting branch and the receiving branch.

a. Transmitting Branch. The *transmitting branch* of the *carrier channel* of a four-wire terminal consists of a hybrid arrangement (common to both branches), a pad, a low-pass filter, a modulator, an oscillator (common to both branches) and a band-pass filter. In operation, voice-frequency signal from the adjacent voice-frequency ringer enters the *hybrid arrangement.* The hybrid arrangement permits the signal to go to the *pad.* The pad attenuates the signal to a predetermined value, making the voice-signal voltage that reaches the modulator considerably less than that from the oscillator. From the pad, the voice signal enters the *transmitting low-pass filter.* There all voice frequencies above 2,800 cycles are rejected, limiting the voice-frequency band to a desired small bandwidth (2,600 cycles) permitting good communication. From the transmitting low-pass filter the voice-frequency band enters the *modulator,* where it is combined with the carrier-frequency signal from the oscillator. The modulator output contains (along with other frequencies) the upper and lower sidebands. The *band-pass filter* that follows the modulator permits either the upper or lower sideband (one only) to pass to the input of the *common transmitting amplifier.* The output of the voice-channel transmitting branch also is connected to the input of the common transmitting amplifier; however, because there is a transmitting low-pass filter at the output of the voice-channel transmitting branch, the carrier-channel sideband does not enter that branch, but enters the common transmitting amplifier only. There its level is raised to the value desired for transmission to the far-end carrier terminal.

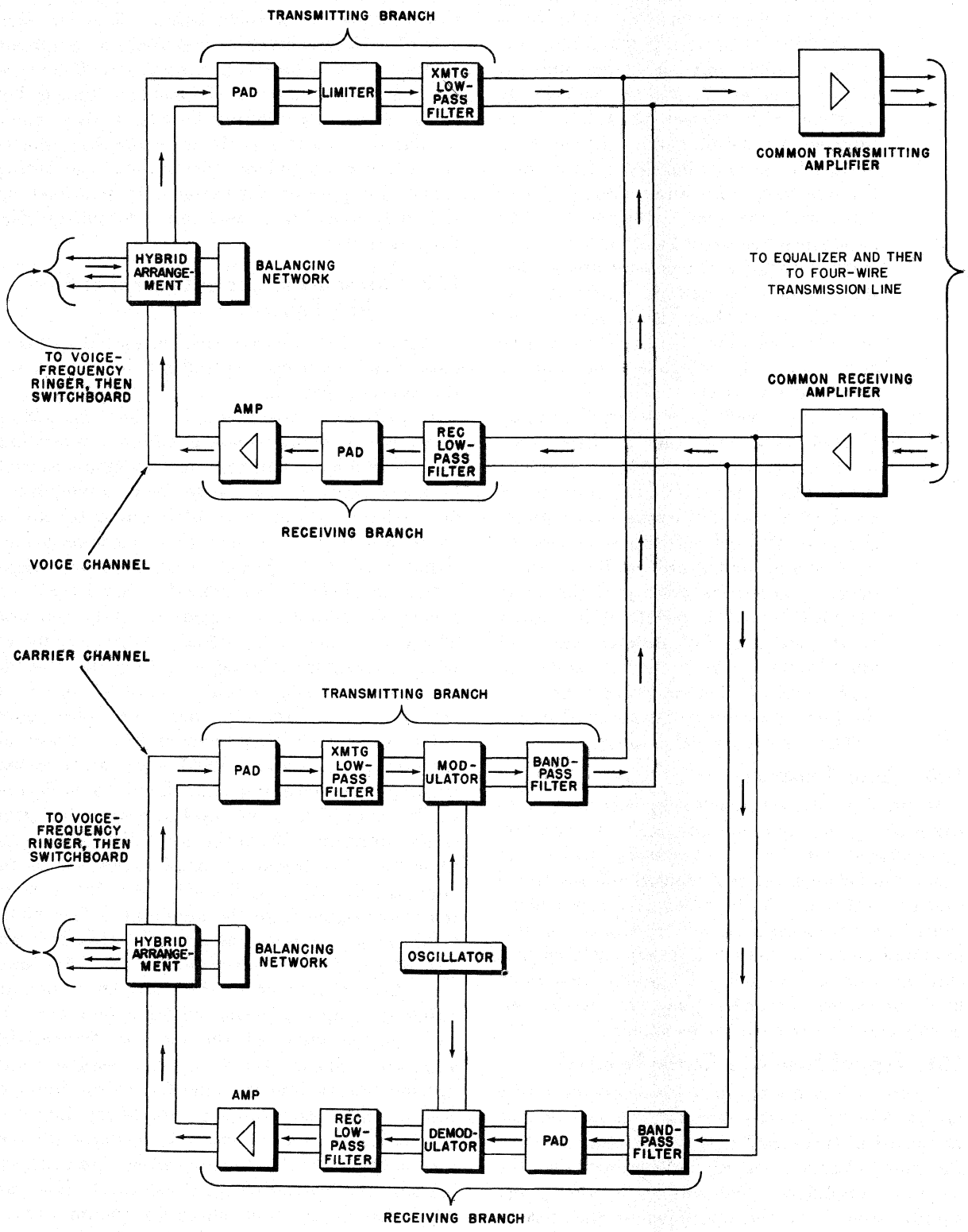


Figure 135. Typical four-wire carrier terminal.

b. Receiving Branch.

(1) The *receiving branch* of the *carrier channel* of a four-wire terminal consists of a band-pass filter, a pad, a demodulator, an oscillator (common to both branches), a low-pass filter, an amplifier, and a hybrid arrangement (common to both branches) (fig. 135). Assume that a signal enters the *common receiving amplifier* from the distant carrier terminal. This signal is in the range of the carrier channel and is at a low level. Passing through the amplifier, the signal level is raised to a predetermined value, which insures proper transmission over the receiving branch. At the output of the common receiving amplifier, the signal sees two paths, through two filters in parallel, which pass different frequency ranges—a low-pass filter in the voice channel, and a band-pass filter in the carrier channel. Because the signal is in the carrier-channel range, it is rejected by the low-pass filter, but it passes readily through the *band-pass filter*. The *pad* following the band-pass filter attenuates the entering signal, making the voice-signal voltage that reaches the *demodulator* considerably less than that from the oscillator. The demodulator reproduces the original voice-signal frequency along with other combinations of the oscillator frequency and the incoming signal. The *receiving low-pass filter* following the demodulator permits only the original voice signal to pass to the amplifier input. The *amplifier* raises the voice-signal level for further transmission through the hybrid arrangement and voice-frequency ringer to the switchboard.

(2) Figure 136 is an illustration of the front view of a typical carrier-channel panel. This unit contains two band-pass filters, two low-pass filters, an oscillator, an amplifier, and an amplifier gain control.

103. Operation of Voice Channel of Four-Wire Terminal

Figure 135 shows that the voice channel consists of two branches, the transmitting branch and the receiving branch.

a. Transmitting Branch. The *transmitting branch* of the *voice channel* of a four-wire terminal consists of a hybrid arrangement (common to both branches), a pad, a limiter, and a transmitting low-pass filter. It differs from the carrier channel in that it contains a limiter, but no modulator, oscillator, or band-pass filter. A voice-frequency signal from the adjacent voice-frequency ringer enters the *hybrid arrangement*. The hybrid arrangement permits the signal to go to the *pad*. There the signal is attenuated, because it is desired to keep all channel signals to the common transmitting amplifier at approximately the same level, to keep them from overloading the amplifier. The *limiter* which follows the pad reduces ab-

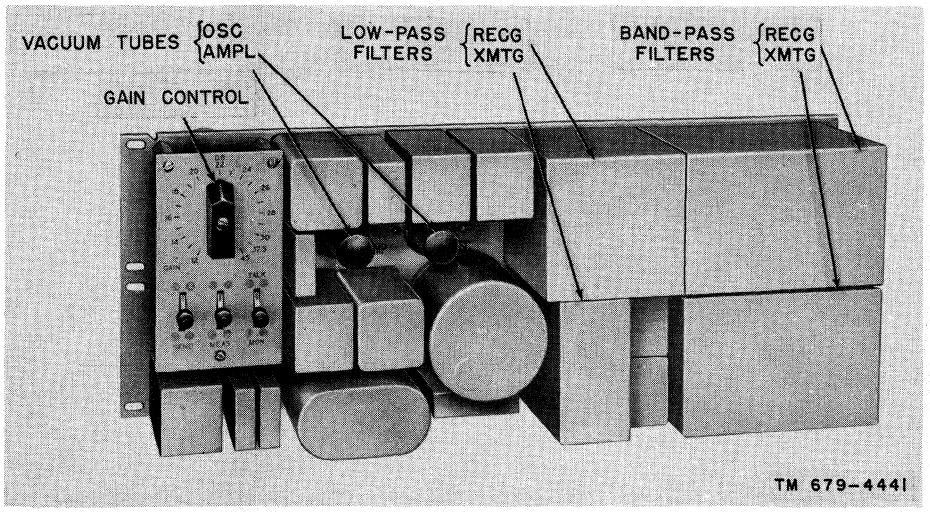


Figure 136. Typical carrier-channel panel, front view.

282463 O—54——8

normally large peak values which would overload the common transmitting amplifier. From the limiter the voice signal enters the *transmitting low-pass filter*. Here all voice frequencies above 2,800 cycles are rejected, limiting the voice-frequency band to a desired small bandwidth (2,600 cycles) permitting good communication. This small bandwidth enters the common transmitting amplifier. The output of the carrier-channel transmitting branch is connected to the input of the *common transmitting amplifier* also; however, because there is a band-pass filter at the output of the carrier-channel branch, the voice signal does not enter that branch, but enters the common transmitting amplifier only. There its level is raised to the value desired for transmission to the far-end carrier terminal.

b. Receiving Branch.

(1) The *receiving branch* of the voice channel of a four-wire terminal consists of a receiving low-pass filter, a pad, and an amplifier (fig. 135). It differs from the carrier channel in that it does not contain a band-pass filter, a demodulator, or an oscillator. Assume that a signal enters the *common receiving amplifier* from the distant carrier terminal after passage through the equalizer. This signal is in the range of the voice channel and is at a low level. Passing through the amplifier, the signal level is raised to a predetermined value—the same value as for a signal in the carrier-channel band. At the output of the common receiving amplifier, the signal sees two paths, through two filters in parallel, which pass different frequency ranges—a receiving low-pass filter in the voice channel, and a band-pass filter in the carrier channel. Because the signal is in the voice-channel range, it is rejected by the band-pass filter; but it passes readily through the *receiving low-pass filter*. The *pad* following the low-pass filter attenuates the entering signal, to keep it from over loading the amplifier which follows and causing distortion. The *amplifier* raises the voice-signal level for further transmission through the hybrid arrangement and voice-frequency ringer to the switchboard.

(2) Figure 137 is a front view of a typical voice-channel panel. This unit contains two low-pass filters, an amplifier, and an amplifier gain control.

(3) Figure 138 is an illustration of the front view of a typical four-wire carrier terminal. This terminal contains a common amplifier panel, an equalizer panel, one voice-channel panel, and three carrier-channel panels. The unit allows the simultaneous transmission of four different two way conversations.

104. Typical Two-Wire Carrier Terminal

A *two-wire carrier terminal* is the same as a four-wire terminal except for additional equipment which converts from four-wire operation to two-wire operation. The additional equipment may be part of the two-wire terminal, or it may be a

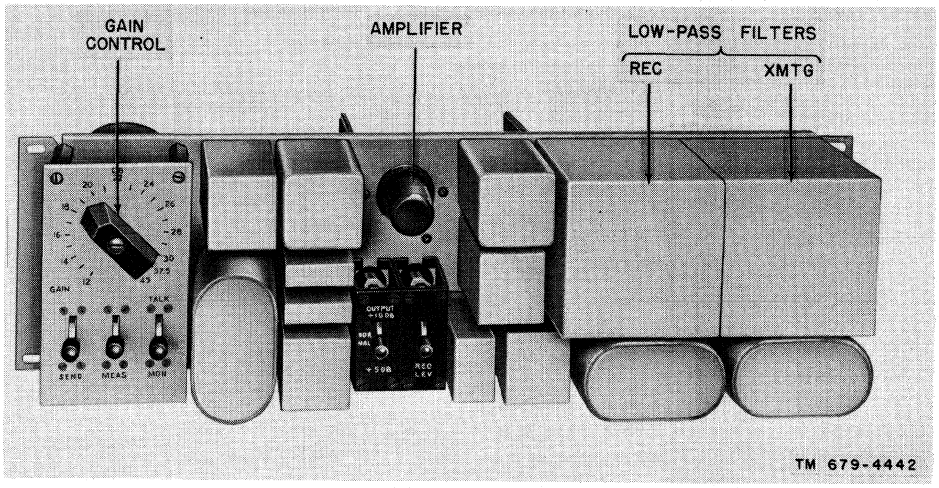


Figure 137. Typical voice-channel panel, front view.

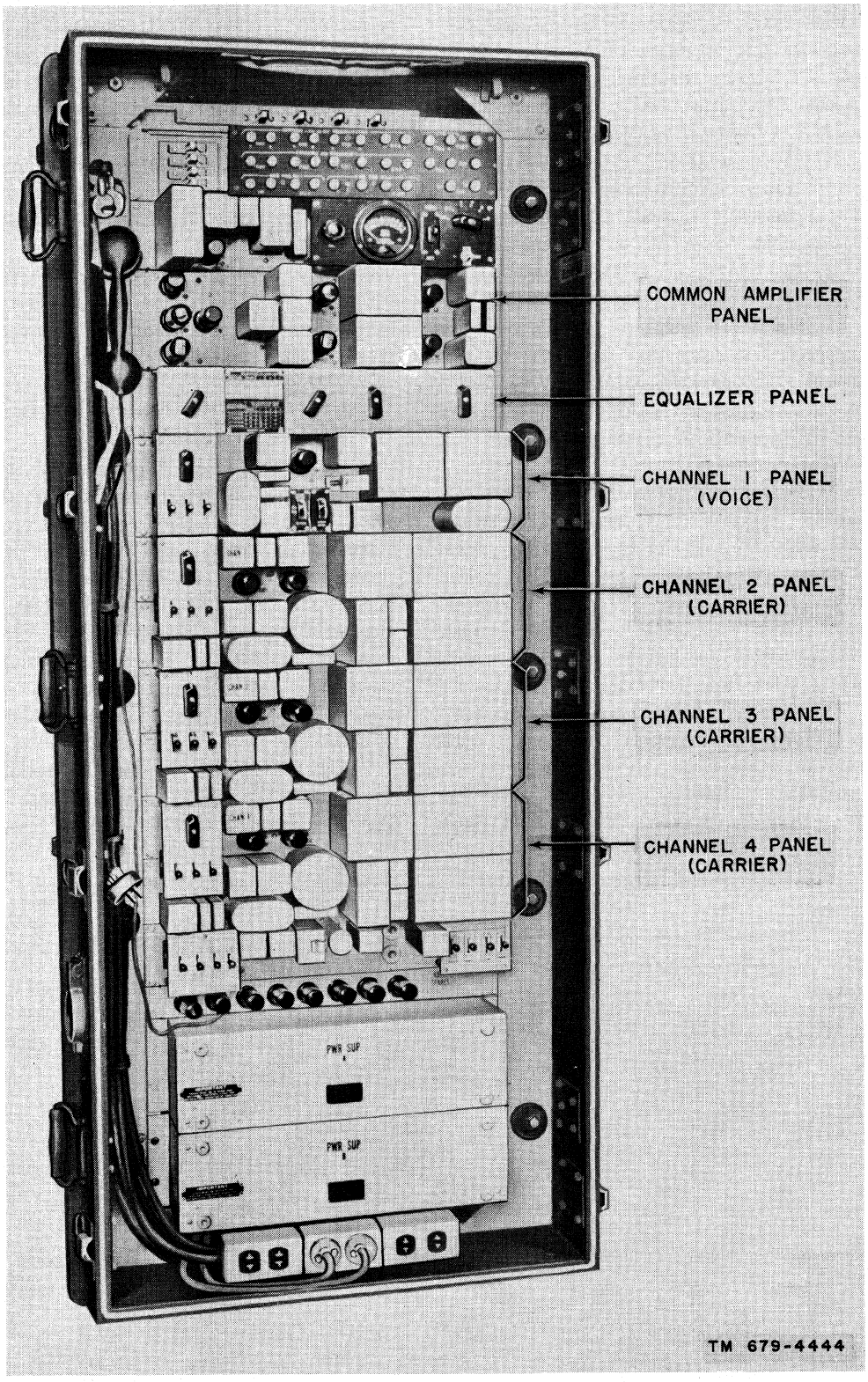


Figure 138. Typical four-wire carrier terminal, front view.

separate unit. This equipment changes the frequency range for one direction of transmission.

a. Figure 135 shows in block form a typical four-wire carrier terminal. Figure 139 shows in block form the *additional equipment* required to convert that four-wire terminal into a two-wire terminal, and shows the manner in which the additional equipment is inserted at both the West and East terminals. Imagine the 4 wires at the right of figure 135 connected to the 4 wires at the left of figure 139. That will give a picture of the arrangement of a West terminal permitting two-wire operation. Imagine the 4 wires at the right of figure 139 properly connected to the 4 wires of a four-wire East terminal, and the arrangement of the components of the entire system will be comprehended.

b. In the West terminal (as in the East terminal) the additional equipment is inserted between the common amplifiers of the four-wire terminals and the two-wire transmission line (fig. 139). The line is connected at the West and East terminals to *directional filters* (transmitting and receiving). The directional filters are band-pass filters which separate the sending and receiving paths at each terminal. (Theoretically, hybrid arrangements may be used to perform this function; practically, over a wide range of frequencies, they give very poor balance, and so are not used.) Note that the additional equipment is not the same at each terminal. At the West terminal it consists of a transmitting directional filter, a receiving directional filter, an amplifier, a group demodulator, and a demodulator filter. At the East terminal the additional equipment consists of a transmitting directional filter, a receiving directional filter, and a group modulator. The West transmitting directional filter and the East receiving directional filter are identical, and the East transmitting directional filter and the West receiving directional filter are identical. The two directional filters at one terminal are not identical, however, and do not pass the same band of frequencies.

105. West-to-East Transmission in Two-Wire Carrier System

a. In *West-to-East transmission* (fig. 139) the output band of frequencies of the *West-terminal common transmitting amplifier* is sent to the *West transmitting* directional filter. Passing readily through the filter, it sees 2 paths—1, through the West receiving directional filter, and the other, out onto the transmission line. Since the two directional filters at a terminal do not pass the same band, the West receiving directional filter rejects this band, and it goes out onto the *transmission line* to the East terminal.

b. Upon entering the East terminal, two paths again are seen by the transmitted band of frequencies, one through each of the East directional

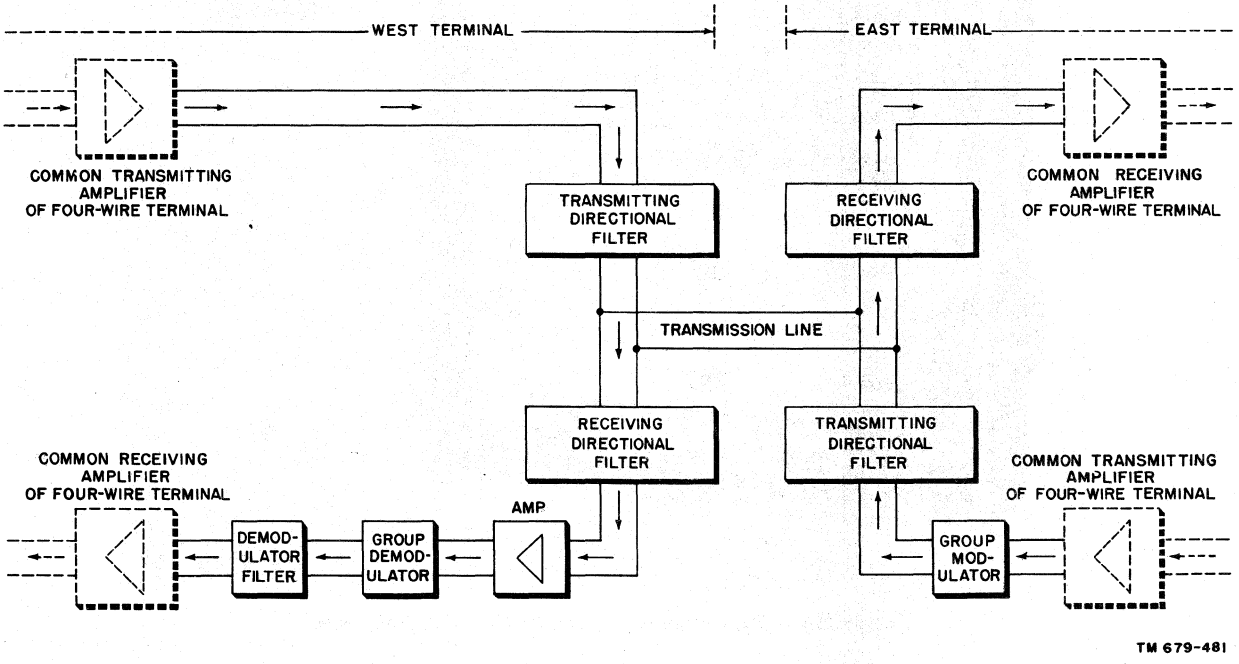


Figure 139. Additional equipment for conversion of four-wire terminal to two-wire terminal.

filters (fig. 139). Since the two directional filters at a terminal do not pass the same band, and since the West transmitting directional filter and the East receiving directional filter are identical, the East transmitting directional filter rejects this band, and it passes readily through the *East receiving directional filter* to the *East-terminal common receiving amplifier.*

106. East-to-West Transmission in Two-Wire Carrier System

East-to-West transmission differs from West-to-East transmission. In the West-to-East direction, filter action only is used; in the East-to-West direction, modulation and demodulation are employed, as well as filter action.

a. In *East-to-West transmission* (fig. 139) the ouptut of the *East-terminal common transmitting amplifier* is sent to the *group modulator.* The *group modulator* modulates a new carrier with several incoming channel frequency bands. All of the channel frequencies are superimposed on the new higher carrier frequency whose value is such that the new lower sideband is well above any of the input channel frequencies. Only 1 of the 2 sidebands in the group modulator output is used (ordinarily) for further transmission. Assume that the lower sideband is used. The East transmitting directional filter permits only this new lower sideband to pass for further transmission. At the ouput of the filter this band sees two paths—one, through the East receiving directional filter, and the other, out onto the transmission line. Since the two directional filters at a terminal do not pass the same band, the East receiving directional filter rejects this band, and it goes out onto the *transmission line* to the West terminal.

b. Upon entering the West terminal, 2 paths again are seen by the transmitted band of frequencies, 1 through each of the West directional filters (fig. 139). Since the two directional filters at a terminal do not pass the same band, and since the East transmitting directional filter and the West receiving directional filter are identical, the West transmitting directional filter rejects this band, and it passes readily through the *West receiving directional filter,* to the *amplifier.* Here the signal level is raised to the desired value for demodulation. The *group demodulator* which follows has the same carrier frequency as the East-terminal group modulator; therefore, the combining of the transmitted band of frequencies and the local carrier frequency in the group demodulator reproduces the original East-terminal channel frequencies along with other, undesired, frequencies. The *demodulator filter* permits only the original channel frequencies to pass through to the West-terminal common receiving amplifier.

107. Control and Alarm Circuits

In carrier operation, certain precautions and additional circuits commonly are used. There may be means for the automatic regulation of amplifier gain; there are circuits for the summoning of attendants when open lines occur (alarm systems), and there must be alining procedures for setting and maintaining the frequency of oscillators.

108. Control of Oscillator Frequency

In telephone carrier systems, the transmitting terminal employs a certain carrier frequency on a particular channel for modulation purposes, and the receiving terminal employs the same carrier frequency for demodulation. Since the channel carrier frequency is suppressed in single-sideband suppressed-carrier systems, no carrier frequency arrives at the receiving end for demodulation purposes. This requires that the carrier frequency be reinserted for demodulation. Any difference in carrier frequencies used for modulation and demodulation will appear in the output of the receiving terminal. For example, assume that a voice frequency of 1,000 cycles enters a modulator in the transmitting terminal, and that the reinserted carrier frequency of the corresponding channel demodulator deviates 10 cycles from that carrier frequency. At the output of the receiving terminal, a voice frequency of 1,010 or 990 cycles will appear, depending on the direction of the 10-cycle deviation. This condition is prevented by the following procedure for alining corresponding channel oscillators:

a. Alinement of Voice-Frequency Oscillator. The procedure for *alining corresponding voice-frequency oscillators* of distant carrier terminals uses 2 voice-frequency oscillators, 1 at each terminal, both (usually) of 1,000 cycles. It is important for both oscillators to operate at exactly the same frequency, but it is not necessary that the frequency of oscillation be exactly 1,000 cycles. The output of one 1,000-cycle oscillator is sent to the distant terminal over the voice channel. Since the voice channel does not contain modulators or demodulators, the signal frequency received is exactly the one transmitted. At the receiving

terminal, this 1,000-cycle signal and the one from the local corresponding voice-frequency oscillator are placed across a *db meter*. If the frequencies differ slightly, the meter needle moves back and forth slowly; if they differ greatly, the needle vibrates rapidly; the *rate* at which the meter needle moves back and forth corresponds to the difference in frequency between the two voice-frequency oscillators. The frequency of the local oscillator must be varied until the movement is slowed down to one representing a few cycles per minute, or to a stop. In this way, two distant voice-frequency oscillators are set to operate at the same frequency. Once the voice-frequency oscillators are alined, the corresponding channel oscillators can be alined.

b. Alinement of Channel Oscillator. The procedure for *alining corresponding channel oscillators* is similar to that for alining the voice-frequency oscillators over the voice channel. At the transmitting terminal, the output of the 1,000-cycle voice-frequency oscillator is placed across the input of the carrier channel to be alined. At the receiving terminal, the output of the corresponding channel and the output of the local voice-frequency oscillator are placed across the *db meter*. The meter needle indicates any difference of frequency. When a difference is indicated, the carrier-channel oscillators are not operating at the same frequency; therefore, the frequency of the local channel oscillator must be varied until the movement of the needle is slowed down to one representing a few cycles per minute, or to a stop. Thus, two corresponding channel oscillators are set to operate at the same frequency. This procedure is continued until all the corresponding channel oscillators are alined.

c. Synchronization of Oscillators. Once the channel oscillators are alined, *continuous synchronization* may, or may not, be used. If continuous synchronization is not used, periodic oscillator alinements must be made. Continuous synchronization is obtained by means of a *pilot frequency*, which is a continuous signal, not containing conversation intelligence, sent from one terminal to the other. This pilot frequency locks the corresponding carrier-channel oscillators in synchronization.

109. Regulation of Amplifier Gain

Because of the long transmission lines of carrier systems, small attenuation changes in line sections—such as are caused by temperature variations, for example—are cumulative to the point where they are appreciable. These changes are compensated by regulation of the gain of the common receiving amplifier. The regulation is effected by means of an automatic amplifier-gain control circuit and a pilot frequency. The pilot frequency from a distant terminal causes the automatic control circuit to vary the level of the output of the local common recieving amplifier. The pilot frequency is not in the range of the conversation-intelligence-carrying frequencies.

a. Description of Automatic Amplifier-Gain Control Circuit. Figure 140 shows in block form an *automatic amplifier-gain control circuit.* This circuit consists of two branches—the all-frequency branch and the pilot-frequency branch. The all-frequency branch consists of the motor-driven

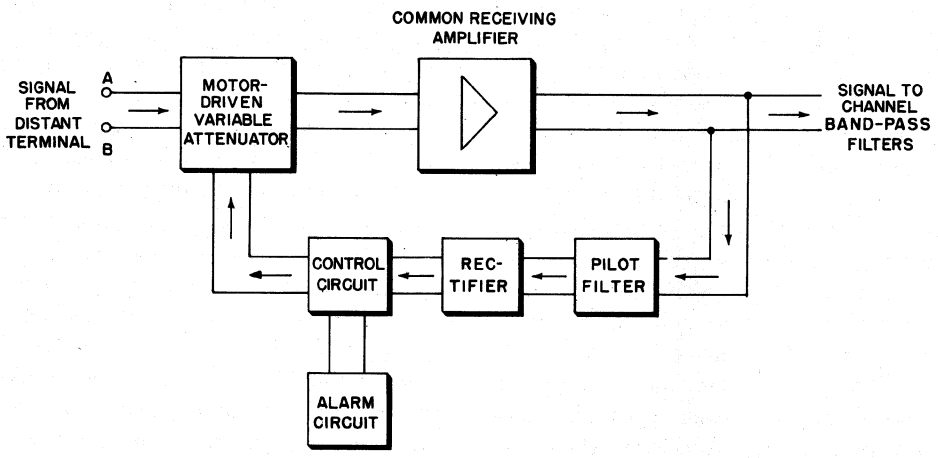


Figure 140. Automatic amplifier-gain control circuit.

variable attenuator and the common receiving amplifier. The pilot-frequency branch consists of the pilot filter, rectifier, control circuit, and alarm circuit. The amplifier is of the fixed-gain type which uses a variable attenuator to control the level of its output.

b. Operation of Automatic Amplifier-Gain Control Circuit.

(1) All conversation-intelligence-carrying frequencies and the pilot frequency from the distant terminal enter at terminals A–B (fig. 140). All of these frequencies pass through the *variable attenuator* to the *common receiving amplifier*. Here the level of all the frequencies is raised to a predetermined value. There are 2 paths at the amplifier output—one path to the channel band-pass filter, the other to the pilot filter. All of the conversation-intelligence-carrying frequencies are accepted by their respective *channel band-pass filters*, but are rejected by the pilot filter.

(2) The pilot frequency, not being in the range of the conversation-intelligence-carrying frequencies, is rejected by the channel band-pass filters, but is accepted by the *pilot filter* (fig. 140). After passing through the pilot filter, the pilot frequency is rectified in the *rectifier*. The rectified current flows through the windings of d-c relays in the control circuit.

(3) While the pilot frequency remains at a predetermined level, the control circuit remains inoperative. A variation of the pilot-frequency level indicates a change in the attenuation of the transmission line. A small variation (usually ± 0.5 *db*) of the pilot-frequency causes the control circuit to operate the motor-driven variable attenuator. The attenuator change is such as to restore the pilot-frequency to its predetermined level—for a $+0.5$-*db* variation (for example) the attenuator *increases* the attenuation in the all-frequency branch; for a -0.5-*db* variation, the attenuator *decreases* the attenuation in the all-frequency branch.

(4) A large variation ($+3$ *db* or -5 *db*) of the pilot-frequency level, present for a certain time interval (usually fixed at 20 to 30 seconds), causes the control circuit to operate the alarm circuit. This is an indication that this large attenuation change cannot be corrected automatically. Should the transmission line become open-circuited, so that current of pilot frequency ceases to enter the receiving terminal, the control circuit, after the fixed time interval, operates the alarm circuit.

(5) The *alarm circuit* (fig. 140) gives a visual or an audible alarm signal, or both. By the alarm signal an attendant is informed that the line is open-circuited, or that manual correction is needed to restore the pilot frequency to its predetermined value.

110. Cable Alarm

In carrier systems, it is essential that failure of the cable be indicated immediately. When the systems use automatic amplifier-gain control circuits, an alarm is given when the line is open-circuited, but when they do not, immediate indication usually is obtained through use of auxiliary equipment and a simplex circuit. The circuit that performs this task is the cable-alarm circuit.

a. Description of Cable-Alarm Circuit. Figure 141 illustrates a *cable-alarm circuit* as used in a four-wire carrier system. (It can be used in a two-wire carrier system, as well.) The figure shows the East terminal, the transmission line, the West terminal, and the alarm circuit. For simplicity, only the carrier-terminal input connections from the transmission line are shown. At each terminal, each pair of line wires is connected to a repeat coil. Only one pair of line wires is simplexed for the alarm circuit. Again, for simplicity, the auxiliary alarm-circuit equipment is shown outside the carrier terminals, not contained inside. At each terminal, this equipment consists of a d-c source (battery), a d-c relay, a key, a ballast lamp, an a-c source, and an a-c buzzer.

b. Operation of Cable-Alarm Circuit.

(1) At the West terminal (fig. 141) the battery is grounded and a d-c path is present through the ballast lamp, key C, relay A, the repeat coil, and the transmission-line pair to the East terminal, where the d-c path is continued through the repeat coil, relay B, and key D to ground. Because the d-c path is closed through ground, direct current flows in this path, and relays A and B are oper-

ated as shown. The ballast lamp is used to insure against overloading the relay windings.

(2) With relays A and B operated (fig. 141), contacts 1 and 2 are open and no alternating current reaches the a-c buzzers from the a-c sources at the terminals. Should the simplexed-line pair become open-circuited, relays A and B no longer would be energized, contacts 1 and 2 would close, and this would cause alternating current to flow through the a-c buzzers from the a-c sources at the terminals. Thus, an audible alarm would be given at each terminal, indicating that the transmission line had become open-circuited. On the other hand, a grounded conductor or bad leakage to ground may cause the alarm circuit to become inoperative, even though transmission over the system is possible.

(3) Note the positions of keys C and D. At the West terminal, key C connects relay A through the ballast lamp to the battery; at the East terminal key D connects relay B directly to ground. These are the correct key positions for proper alarm-circuit operation. If key D should be operated, key C remaining in the position shown, the distant battery supplies will be in opposition, no current will flow to energize the relays, and both buzzers will sound. If key C should be operated, key D remaining in the position shown, there will be no battery source in the d-c circuit to energize the relays, and again both buzzers will sound. If either key C or D is operated in error, the a-c buzzers are always connected to the a-c sources through made contacts 1 and 2, and will sound until the error is corrected. For proper operation, one key must connect its adjacent relay directly to ground, and the other key must connect its adjacent relay through the ballast lamp to the battery.

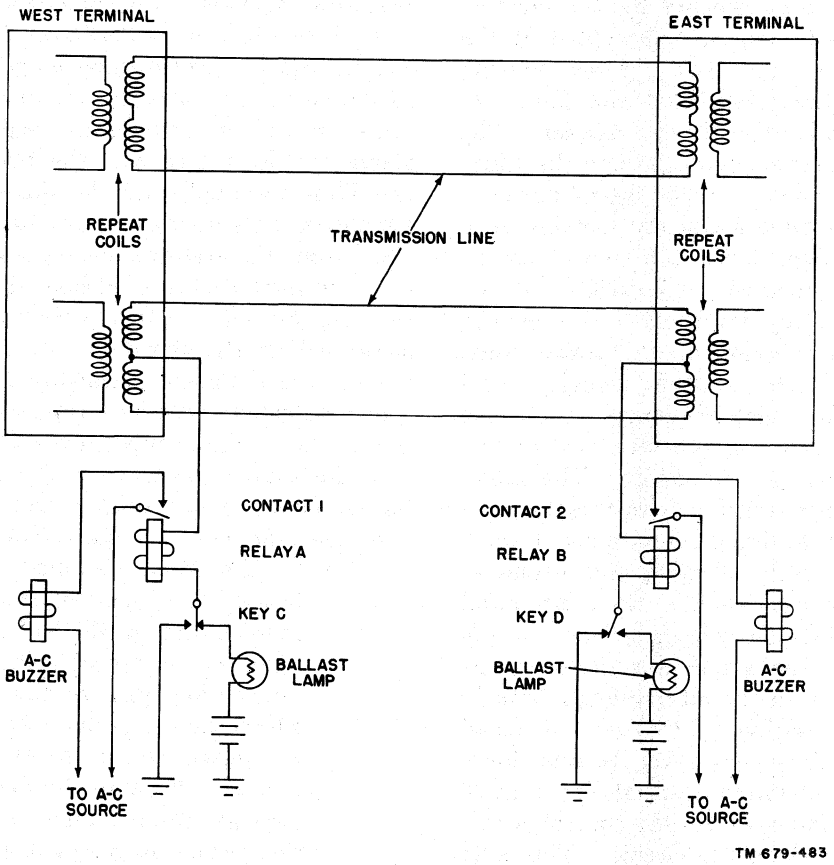


Figure 141. Cable-alarm circuit.

111. Carrier-Supply Alarms

In carrier systems, a *carrier-supply alarm circuit* may be used to indicate an important change in carrier voltage, and another may be used to indicate an appreciable shift in carrier frequency. One or both of these alarm circuits may be employed in one carrier system at the same time.

a. Carrier-Voltage Alarm Circuit.

(1) The circuit that indicates a change in the carrier voltage is called a *carrier-voltage alarm circuit.* Figure 142 shows the circuit in the form of a simplified block diagram. It consists of a rectifier, a control circuit, and an alarm circuit, and its input lies across the channel-oscillator output, in parallel with the modulator and demodulator.

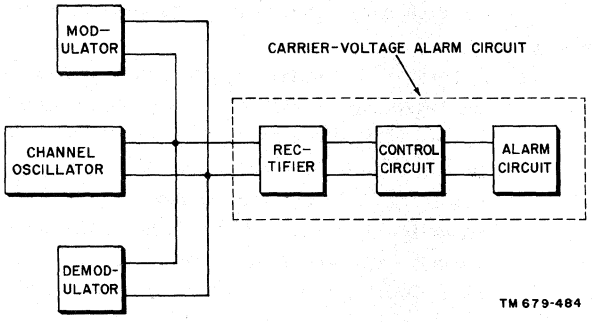


Figure 142. Carrier-voltage alarm circuit.

(2) The oscillator signal enters the rectifier, and this rectified current flows through the windings of d-c relays in the control circuit (fig. 142). While the output of the channel oscillator remains at a predetermined value, the control circuit remains inoperative. A variation in the output of the oscillator causes a variation in the amount of rectified current flowing through the windings of the d-c relays in the control circuit. When the variation reaches a predesignated value, the relays are activated, and the control circuit operates the alarm circuit. The alarm may be visual or audible, or both.

b. Carrier-Frequency Alarm Circuit.

(1) The circuit that indicates a shift in the carrier frequency is called a *carrier-frequency alarm circuit.* Figure 143 shows the circuit in the form of a simplified block diagram. It consists of a filter, a rectifier, a control circuit, and an alarm circuit, and its input lies across the channel-oscillator output, in parallel with the modulator and demodulator.

(2) The filter has a very narrow band-pass with the channel-oscillator frequency in the center. The oscillator signal that passes through the filter enters the rectifier, and this rectified current flows through the windings of d-c relays in the control circuit. While the oscillator frequency is correct, a sufficient amount of signal passes through the filter and rectifier to keep the control circuit in operation. A shift in the frequency of the oscillator causes a lessening of the amount of rectified current flowing through the windings of the d-c relays in the control circuit. When the shift reaches a predesignated number of cycles above or below the desired frequency, too little of the signal current passes through the filter and rectifier to the windings of the d-c relays to keep the relays activated; consequently, the relay contacts are released, and the control circuit operates the alarm circuit. The alarm may be visual or audible, or both.

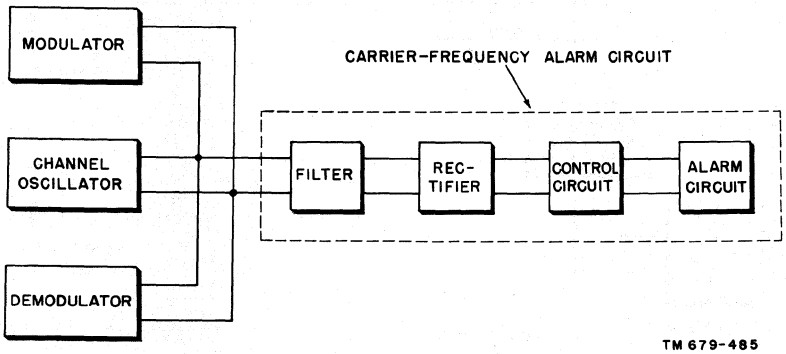


Figure 143. Carrier-frequency alarm circuit.

112. Carrier-Frequency Repeaters

In long-distance carrier systems, carrier-frequency repeaters are used to compensate for transmission-line attenuation in the same manner as voice-frequency repeaters are used in long voice-transmission systems. Carrier-frequency repeaters consist of amplifiers and attached networks. They are used in both two-wire and four-wire carrier systems.

a. Description of Two-Wire Repeater. Figure 144 is a block diagram of a *two-wire carrier-frequency repeater.* It consists of two amplifiers and four directional filters. The amplifiers transmit in opposite directions. The directional filters are used (instead of hybrid arrangements (par. 104*b*) to separate the sending and receiving paths. The West receiving and East transmitting directional filters are identical, and the East receiving and West transmitting directional filters also are identical. The filters adjacent to the West line are not identical, however, nor are those adjacent to the East line. This two-wire repeater can be inserted in the transmission line between the carrier terminals shown in figure 139.

b. Operation of Two-Wire Carrier Repeater.

(1) For opposite directions of transmission in a two-wire transmission system, two different frequency bands are employed (pars. 104–106). Assume that in figure 144 a signal is entering the repeater from the West line. The West transmitting directional filter rejects this frequency band, but the *West receiving directional filter* readily passes it to the *W–E amplifier,* where it is amplified for further transmission and goes through the *East transmitting directional filter* out onto the *East line.* The East receiving directional filter keeps this frequency band from entering the E–W amplifier input, thus preventing feedback, which may cause singing.

(2) Similarly, a signal entering the repeater from the East line, because of its frequency band, passes only through the *East receiving directional filter,* the *E–W amplifier,* and the *West transmitting directional filter,* and so goes out onto the *West line.*

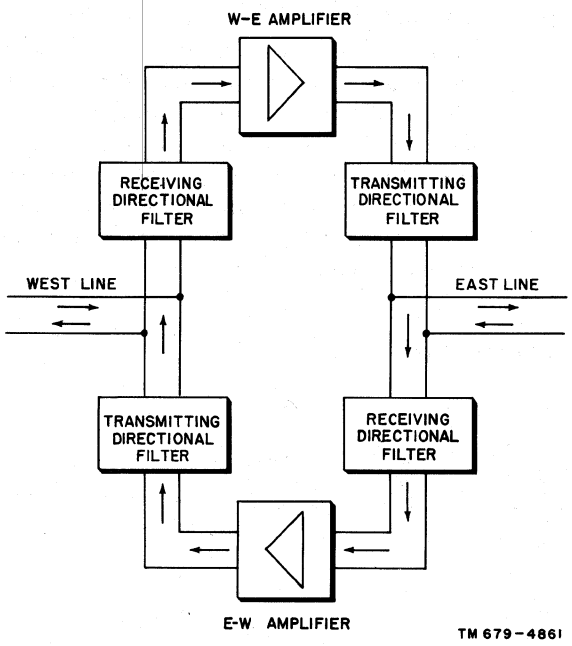


Figure 144. Two-wire carrier-frequency repeater.

c. Description and Operation of Four-Wire Carrier Repeater.

(1) A four-wire carrier frequency repeater is the same as a two-wire carrier repeater, except for the directional filters of the latter which are not necessary. A diagram of a four-wire carrier repeater is shown in figure 145. The four-wire carrier repeater is less complex than the two-wire carrier repeater, and is widely used.

(2) Figure 146 is an illustration of the front view of a typical four-wire carrier repeater. This unit contains a common amplifier panel, two equalizer panels, and a telephone handset. The common amplifier panel consists of two two-stage amplifiers, one for each direction of trans-

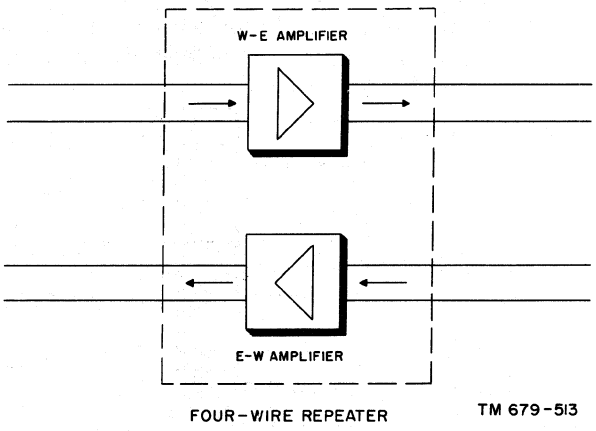


Figure 145. Four-wire carrier repeater

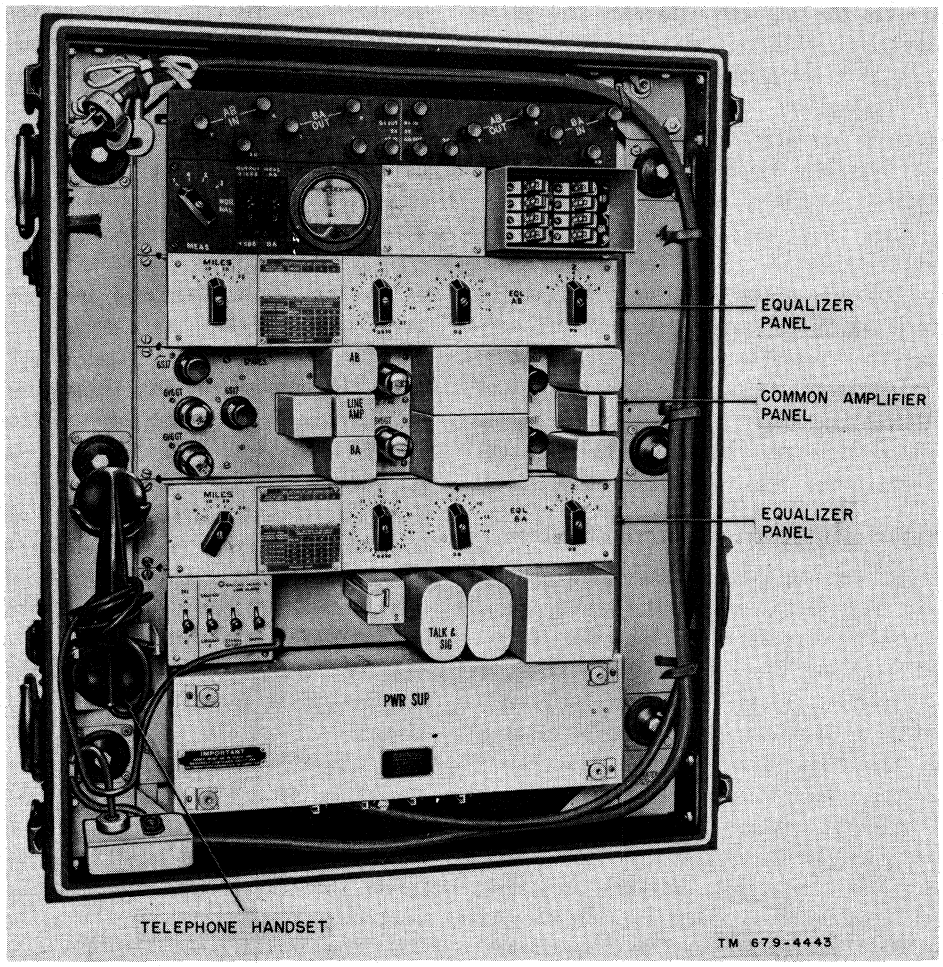


Figure 146. Four-wire carrier repeater, front view.

mission. Two equalizer panels are needed to equalize both line sections that terminate at the repeater. The handset is connected in such a manner that it can receive and transmit over the voice channel, making it possible for the attendant to communicate with other repeater stations or the terminals.

113. Spacing of Repeaters

The problems involved in the *spacing of carrier-frequency repeaters* are similar to those in the spacing of voice-frequency repeaters. Their solutions are embodied in easy-to-use repeater-spacing specifications covering each type of carrier system.

114. Combination of Two or More Systems

Military carrier systems are combined on open-wire lines only; that is, they are combined by placing two or more carrier systems on the same pole line. In arrangements of this type, it is advantageous to have the location of the carrier repeaters of the several carrier systems located at the same points, since maintenance is centralized and the possibility of crosstalk, because of excessive differences in level, is reduced.

a. Frequency Coordination in Combined Systems. It is desirable that transmission of like carrier-frequency bands be in the same direction. When this is arranged the frequencies are said to be *coordinated. Frequency coordination* tends to prevent the high near-end crosstalk that would occur if currents of the same frequency were at high level on one page of wires and at a low level on another pair of wires, at the same point on the line. Adjacent pairs are used for the same direction of transmission, so that the difference in dbm level between them is minimum. When bands of like frequency must be transmitted in opposite directions, their wire pairs are kept as far apart

as is practicable; the repeater spacings and gains should be reduced. These precautions tend to keep crosstalk at a tolerable level.

b. Level Coordination in Combined Systems. When different types of systems are operated on the same pole line, the levels of their outputs are made approximately the same on all systems for a given frequency range on the line. This usually gives optimum results in the matter of crosstalk. When a branch line is joined to a main line at an intermediate point in the main-line repeater section, or when a repeater or carrier terminal is placed at an intermediate point, readjustment of system levels may be required to equalize the levels at the intermediate point. Means of making such adjustments are included in the carrier equipment.

115. Integrated Multichannel Carrier Systems

a. The carrier equipment now being developed is designed to fit into the integrated signal communication system, having units designed to operate on the building block principle. Thus, a basic unit, or a group of units, can be built into systems as large or as small as required by adding together the necessary number of basic units or blocks. For example, a military unit may require a single set of telephone carrier equipment, and this equipment will provide four voice channels. Assume that the situation then becomes such that more carrier equipment is necessary. With the building block integrated system, it is necessary only to parallel additional basic units to the original installation. The expansion in facilities is accomplished by adding basic units, and it is not necessary to remove the old unit and replace with a larger one.

b. The basic carrier unit now being procured is the newly developed Telephone Terminal AN/TCC–3( ). It is the basic building block for 4-, 12-, and 48-channel systems, and is designed for tactical field use in conjunction with the newly developed type of spiral-four Cable Assembly CX–1064/G, and also for use with radio relay links. With a repeater spacing of 25 miles, the system will provide good communication for a distance of 150 miles.

116. Summary

a Theoretically, a transmission line is capable of transmitting frequencies ranging from zero to infinity. Because of practical considerations, however, an open-wire transmission line, for example, is capable of efficient transmission of frequencies ranging only from zero to approximately 150,000 cycles. The frequency-handling capabilities of cable, not including coaxial cable, is much lower.

b. Reallocation (spacing out) of the voice channel to obtain several facilities over the same transmission line is accomplished by carrier systems. This is known as frequency division multiplexing.

c. A carrier current is an alternating current of a constant predetermined frequency serving to carry the voice-current intelligence superimposed on it.

d. The process of superimposing voice-current intelligence on a carrier current is called modulation, and the device that makes this process possible is called a modulator.

e. The process of separating voice intelligence from a carrier is called demodulation, and the device that makes this process possible is called a demodulator. The process of demodulation is similar to that of modulation.

f. The purpose of each channel of a carrier terminal is to provide a single two-way conversation.

g. When two different frequencies are combined in a modulator, the sum and the difference of the two original frequencies (and their harmonics) are in the output of the modulator. The former is called the upper sideband, the latter the lower sideband.

h. The transmission of one sideband only is called single-sideband suppressed-carrier transmission.

i. In a sense, demodulation is remodulation, for in demodulation one of the two sidebands is combined with the frequency of the original carrier to reconstruct the original voice-transmitted signal.

j. The combination of frequency components in the outputs of associated modulators and demodulators is the same. Also, the circuit arrangements of modulators and demodulators can be the same.

k. The modulators and demodulators in present-day carrier systems widely employ copper-oxide rectifiers called copper-oxide varistors, or, simply, varistors.

l. Varistors often are used, instead of vacuum tubes, because of their simplicity, greater stability, and apparently unlimited life.

m. Copper-oxide varistors, like vacuum-tube diodes, permit current to flow readily in one direction only. The electron current flows readily from the copper to the copper-oxide.

n. Germanium-crystal diodes are replacing cop-

per-oxide rectifiers and vacuum-tube diodes in many applications because of their small size.

o. In telephone carrier systems, copper-oxide varistors and germanium-crystal diodes are used in modulators and demodulators.

p. Two common types of modulator (and demodulator) are the balanced-bridge modulator and lattice-type modulator. In these circuits, the rectifier units operate as carrier-controlled switches; therefore, the carrier voltage must be greater than the voice voltage.

q. The output of the balanced-bridge modulator contains the original voice frequency, the upper and lower side bands, and other frequency components not of any importance. The original carrier frequency is not in the output.

r. The output of the lattice-type modulator contains only the upper and lower sidebands. The carrier and voice signals are suppressed.

s. A circuit used to give a higher loss to some frequencies than it does to others is called an electrical filter, or, more commonly, a filter. Since filter loss varies with frequency, filters must consist of elements that have impedances which vary with frequency.

t. Filters are of four types—low-pass, high-pass, band-pass, and band-stop.

u. The low-pass filter readily permits passage of frequencies from zero up to a predetermined frequency, and attenuates all others.

v. The high-pass filter readily permits passage of frequencies from a predetermined frequency (not zero frequency) up to infinite frequency, and attenuates all others.

w. The band-pass filter readily permits passage of a band of frequencies that lies between two predetermined frequencies (neither of which is zero frequency or infinite frequency), and attenuates all others.

x. The band-stop filter attenuates all frequencies within a specified band, but passes all frequencies above or below that band.

y. To perform filtering effectively, the filter must consist of at least two elements.

z. The pass region is a range of frequencies for which, theoretically, the attenuation is zero. Actually, the attenuation is a small, negligible amount.

aa. The frequency at which attenuation starts to increase rapidly is called the cutoff frequency.

ab. The attenuation region is a range of frequencies for which the attenuation is appreciable.

ac. A simple form of low-pass filter consists of 2 series inductors and 1 shunt capacitor.

ad. A simple form of high-pass filter consists of 2 series capacitors and 1 shunt inductor.

ae. A simple form of band-pass filter consists of a low-pass and a high-pass filter which have overlapping cutoff frequencies.

af. A network which attenuates all frequencies equally is called a resistive pad, or, simply, a pad. A pad consists only of resistive elements.

ag. Symmetrical pads are inserted between matched networks for purposes of attenuation only.

ah. Nonsymmetrical pads are inserted between mismatched networks for the performance of matching as well as for purposes of attenuation.

ai. Four types of pads are the T, the pi, the H, and the square pads.

aj. In a pad, the greater the series resistance, the greater is the attenuation, and the smaller the shunt resistance, the greater is the attenuation.

ak. The T and pi pads are used when one side (the side not containing series resistance) of the attached networks is grounded, or when it is not necessary to have the attached networks balanced to ground.

al. The H and square pads are used when the attached networks are balanced to ground.

am. The attenuation of a transmission line is not the same at all frequencies. The network which adds attenuation, so that the total attenuation is the same for all frequencies, is called an attenuation equalizer, or, simply, an equalizer.

an. An equalizer can be inserted in a circuit at either the sending or the receiving end. It is more commonly inserted at the receiving end.

ao. A network that automatically reduces large peak values is called a *limiter.* The limiters in telephone carrier equipments widely use copper-oxide varistors and germanium-crystal diodes.

ap. Two types of limiters are the directly connected and the magnetically coupled.

aq. Limiters are used in the voice channel of telephone carrier terminals to prevent overloading of the common transmitting amplifier.

ar. A two-wire carrier terminal is the same as a four-wire terminal except for additional equipment which converts from four-wire to two-wire operation.

as. In a two-wire carrier system, directional filters are used to separate the sending and receiving paths.

at. Corresponding channel oscillators of distant carrier terminals have to be alined. This is done

periodically unless a pilot frequency is used to lock the oscillators in synchronization.

au. Transmission-line attenuation is a variable which can be compensated for automatically by means of an automatic amplifier-gain control circuit and a pilot frequency.

av. Through the use of a simplex circuit and auxiliary equipment in a carrier terminal, immediate indication is obtained of an open-circuited cable. This arrangement is called a cable-alarm circuit.

aw. The circuit that indicates a change in the carrier voltage is called the carrier-voltage alarm circuit.

ax. The circuit that indicates a shift in the carrier frequency is called the carrier-frequency alarm circuit.

ay. In long-distance carrier systems, carrier-frequency repeaters are used. There are two-wire and four-wire carrier-frequency repeaters.

az. Military carrier systems are combined on open-wire lines only; that is, two or more carrier systems are placed on the same pole line.

117. Review Questions

a. What is accomplished by means of a carrier system?

b. What is carrier current?

c. What is modulation? What is a modulator?

d. What is demodulation? What is a demodulator?

e. What is the purpose of each channel of a carrier system?

f. What is the upper side band? What is the lower side band?

g. A voice frequency of 1,500 cycles and a carrier frequency of 10,000 cycles are combined in a modulator. What are the side band frequencies produced?

h. A sideband frequency of 10,500 cycles and a carrier frequency of 10,000 cycles are combined in a demodulator. What voice frequency is produced?

i. Where are copper-oxide varistors and germanium-crystal diodes used in carrier equipment?

j. Why are copper-oxide varistors used more widely than vacuum-tube diodes in carrier-system modulators and demodulators?

k. In what direction does the current flow readily in a copper-oxide varistor? In a germanium-crystal diode?

l. Name two types of modulator that use rectifying elements.

m. Draw the circuit for the balanced-bridge modulator. For the lattice-type modulator.

n. Do the outputs of the balanced-bridge modulator and the lattice-type modulator contain the carrier frequency?

o. What is the purpose of a filter?

p. Of what types of elements does a filter consist?

q. What is the purpose of a low-pass filter? Of a high-pass filter? Of a band-filter? Of a band-stop filter?

r. How is cut-off frequency defined?

s. Design a low-pass filter for use in a 600-ohm circuit and a cut-off frequency of 4,000 cycles.

t. Design a high-pass filter for use in a 1,000-ohm circuit and a cut-off frequency of 3,000 cycles.

u. Design a band-pass filter for use in a 500-ohm circuit and a pass region of 6,000 to 9,000 cycles.

v. What is the purpose of a pad?

w. Design a symmetrical T pad having a 15-db loss in a 1,000-ohm circuit, using table III. Check results, using the design equations.

x. Design a symmetrical pi pad having a 12-db loss in an 800-ohm circuit.

y. Design a symmetrical H pad having a 20-db loss in a 600-ohm circuit, using table III.

z. Design a symmetrical square pad having a 20-db loss in a 600-ohm circuit.

aa. Does the attenuation of a transmission line vary with frequency?

ab. What is the purpose of an equalizer?

ac. What is the purpose of a limiter? State one place where it is used in a carrier terminal.

ad. Describe the operation of a directly connected limiter.

ae. Describe the operation of a magnetically coupled limiter.

af. Why is a limiter used in the transmitting branch of the voice channel?

ag. Describe how a four-wire carrier terminal is converted into a two-wire carrier terminal.

ah. What is a pilot frequency?

ai. Describe the operation of the automatic amplifier-gain control circuit.

aj. What is the purpose of the cable-alarm circuit?

ak. What is the purpose of the carrier-voltage alarm circuit?

al. What is the purpose of the carrier-frequency alarm circuit?

am. Describe the operation of a two-wire carrier-frequency repeater.

an. What is the purpose of directional filters?

CHAPTER 5
CARRIER TELEGRAPHY

Section I. INTRODUCTION TO CARRIER TELEGRAPHY

118. Components of a Carrier System

a. Telegraphy is a method of transmitting intelligence by means of electrical impulses in accordance with a previously adopted code. The teletypewriter transmits a series of d-c pulses to a distant point where the receiving unit of the TT (teletypewriter) accepts these signals and converts them to printed copy. Because these signals are d-c, the number of messages that can be transmitted simultaneously over a line is limited to one in each direction. In order to increase the capacity of the line, these d-c signals are converted to carrier telegraph signals which are alternating current. In general, these a-c signals may be treated as speech signals. They can be amplified by a repeater and transmitted over both carrier and radio systems.

b. Figure 147 is a block diagram of a carrier telegraph system. The main components of the system are the teletypewriter station, the loop circuit, the carrier telegraph terminal, and the line or other facility between carrier telegraph terminals. The signals transmitted by the teletypewriter station are d-c. These signals are received by terminal A and converted to a-c signals which are transmitted over the line to terminal B. At terminal B, the a-c signals are converted to d-c signals, which are relayed to the distant teletypewriter station. Transmission in the opposite direction is accomplished in a similar manner.

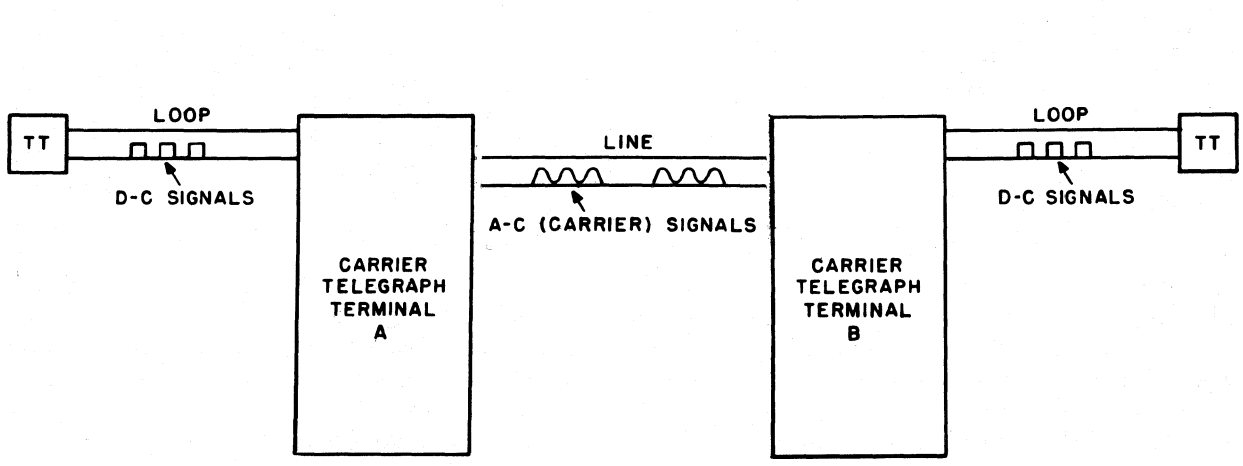


Figure 147. Block diagram of a carrier telegraph circuit.

119. Components at Teletypewriter Station

The teletypewriter station may be a teletypewriter or a teletypewriter switchboard. There is only one channel of communication between each station and the carrier terminal although there may be simultaneous transmission and reception.

120. Loop Circuits

a. Unrepeated Circuits. The circuit between the carrier terminal and the teletypewriter station is the *loop circuit* or d-c extension. When the circuit terminates in a switchboard it is known as a *trunk* (fig. 148). Many local teletypewriters may be connected to the switchboard and plugged into the trunk circuit when communication with the distant switchboard is desired. The circuit layout is very similar to the layouts of telephony, where handsets are connected to a switchboard which terminates trunk circuits.

b. Repeatered Loops. When the teletypewriter station is located at a distance great enough to reduce the signal strength below the value for proper operation, a repeater is inserted into the loop circuit (channel 2, fig. 148). The repeaters renew the signal strength but, in most cases, the signal still contains the distortion that was in the original signal. This distortion again limits the length of the circuit. However, a repeater known as a regenerative repeater may be used. This not only renews but reshapes and retimes the signal as well.

121. Carrier Telegraph Terminal

a. Figure 148 illustrates a carrier telegraph terminal with facilities for four loop circuits.

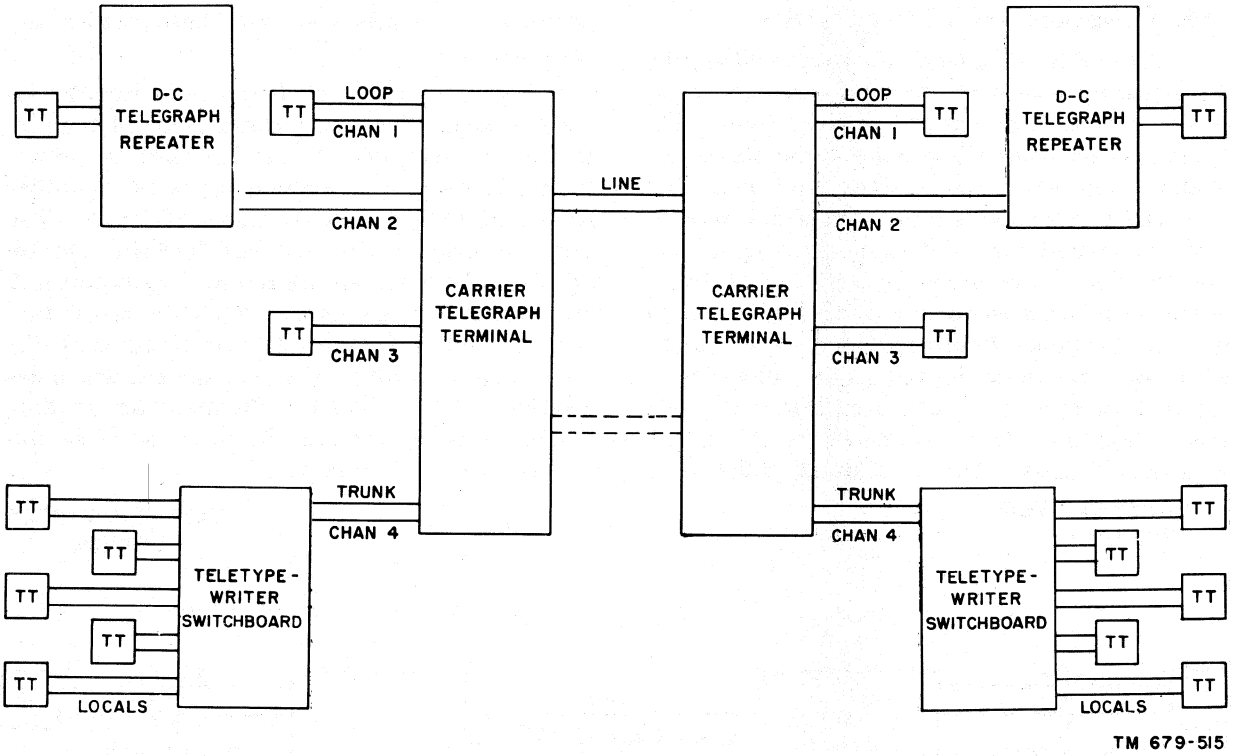


Figure 148. Carrier telegraph terminal and associated circuits.

The number of circuits depends on the design of the terminal. Associated with each loop circuit is a separate set of carrier components (fig. 149). In the terminal illustrated there are four sets of carrier components; therefore, four channels of communication are provided.

b. Each channel has a separate oscillator whose output frequencies differ from those of the other oscillators. D-c signals received from the teletypewriter station connected to channel 1 at carrier telegraph terminal A operate a SEND relay which in turn keys the 1,455-cps oscillator of the

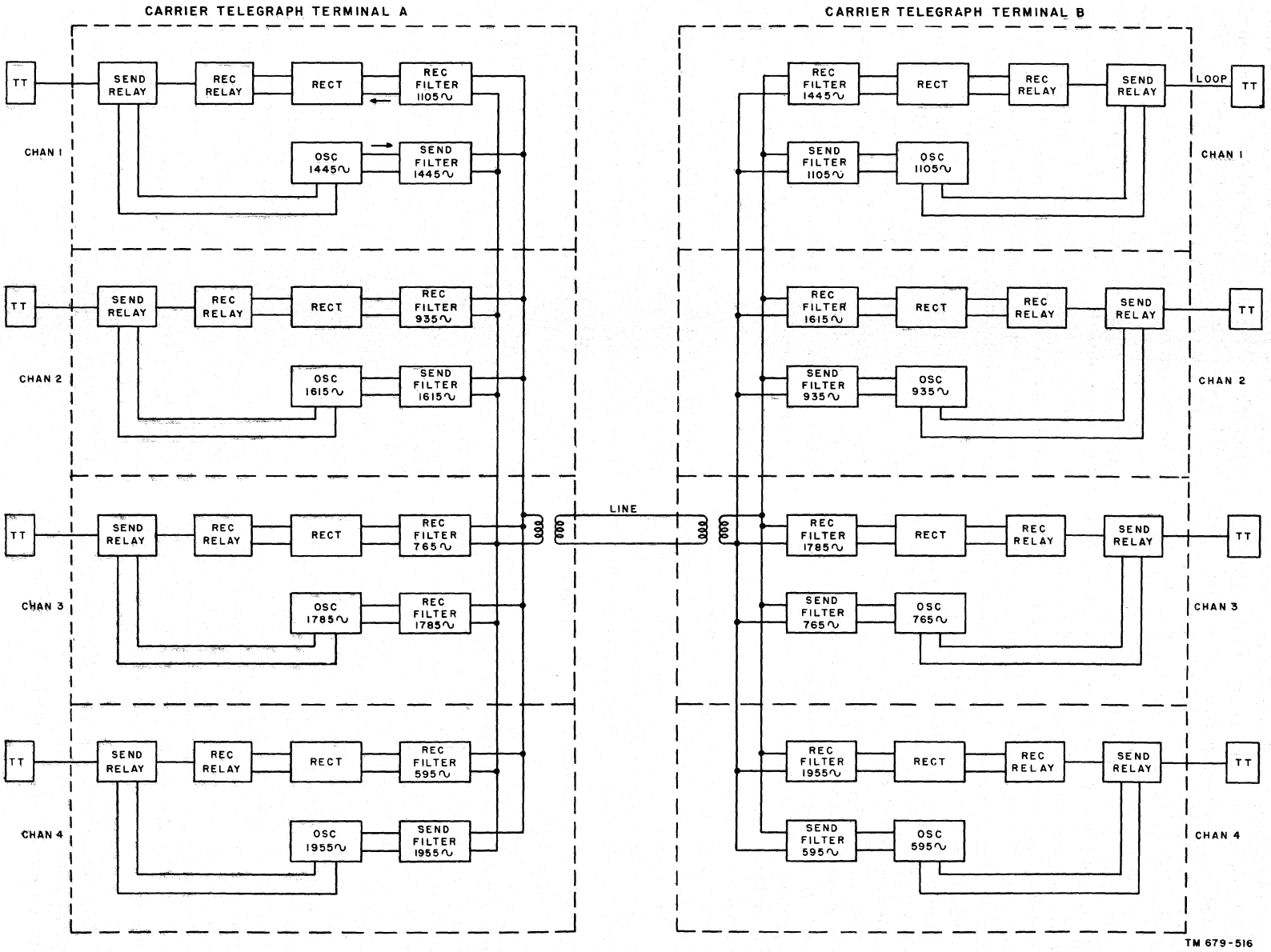


Figure 149. Block diagram of carrier telegraph circuit.

associated channel. This process is called modulation. The signals are then pulses of a-c current. The SEND filter is in the circuit to prevent any unwanted frequencies from reaching the line. D-c signals from the TT connected to channel 2 will cause the SEND relay of channel 2 to key the 1615-cps oscillator. Similarly, the d-c signals from the TT connected to channel 3 keys the 1785-cps oscillator and the TT connected to channel 4 keys the 1955-cps oscillator. The four channel frequencies then are applied to the line. Since the output of each oscillator is at a different frequency, messages may be transmitted simultaneously over the line without interfering with one another. The transmitted signals are received at the B carrier telegraph terminal where the 1,445-cps REC filter of channel 1 will pass only the transmitted frequencies of channel 1 of the A carrier telegraph terminal and reject all others. In this way, channel 1 at A will transmit to and receive from channel 1 at the B terminal. The received a-c signals are rectified and changed to d-c signals. This process is called demodulation. The d-c signals are sent to the teletypewriter, which converts the electrical pulses to printed copy. Similarly, the 1,105-cps REC filter of channel 1 at terminal A will pass only the 1,105-cps frequency transmitted by channel 1 of terminal B.

122. Facilities between Carrier Telegraph Terminals

a. Lines. The facility between the two carrier telegraph terminals illustrated in figure 149 is a two-wire transmission line. The operation may be over a four-wire transmission line, however, as well as over a two-wire line. Four-wire operation may be accomplished by connecting all of the send filters at the A carrier telegraph terminal and all of the receive filters at the B carrier telegraph terminal to the same pair of wires. The SEND filters at the B carrier telegraph terminal and the REC filters at the A carrier telegraph terminal are connected to the second pair of wires.

b. Repeatered Lines. The transmission lines between carrier telegraph terminals attenuate the signals. When the signals are too weak for satisfactory operation, one or more repeaters may be inserted into the circuit.

c. Telephone Carrier System. Usually the frequencies used by the carrier telegraph terminal are in the voice-frequency band. They then may be transmitted over a channel of a *carrier telephone* system. Figure 150 illustrates a voice-frequency telegraph system being operated over one channel (CHAN–3) of a carrier telephone system. The other channels (CHAN–1, CHAN–2, and CHAN–4) of the carrier telephone system are used for speech circuits. In this manner, the message capacity of the overall system has been increased. The bandwidth of a channel of carrier telegraph is narrow. A typical frequency separation between channels is 170 cycles, which allows many channels of carrier telegraph to be transmitted over one telephone channel. To explain the operation of this system, the path of a signal transmitted from the A carrier telegraph terminal to the B carrier telegraph terminal is described as follows: The teletypewriter at the end of the CHAN–1 loop transmits a d-c signal to the A carrier telegraph terminal where it is converted to an a-c signal of 1,445 cycles per second. This a-c signal is transmitted from the line side of the A carrier telegraph terminal to CHAN–3 of the carrier telephone terminal. The CHAN–3 channel oscillator in the carrier telephone terminal puts out an 8,850-cps sine-wave signal which, when combined with the 1,445-cps signal in the modulator, results in the production of upper and lower sideband frequencies. The upper sideband is suppressed by filters and the lower sideband is transmitted. The signal frequency is now 8,850 minus 1,445 or 7,405 cps. This signal is transmitted to the line and to the carrier telephone terminal at B. The 7,405-cps signal is demodulated in the carrier telephone terminal and the signal, now at 1,445 cps, is transmitted to the carrier telegraph terminal. Here it is passed by the REC filter in CHAN–1 of the carrier telegraph terminal, is rectified, and relayed as a d-c signal to the loop and to the teletypewriter.

d. Radio. Teletypewriter transmission by means of radio channels has also come into wide use. This type of transmission is of great value, especially between points that are inaccessible or difficult to reach by wire facilities. Radio may be used over water, long distances of jungle areas, or between two mountain peaks. Figure 150 illustrates this system of transmission. The carrier telegraph terminal may be connected directly to the radio system or it may be connected to a carrier telephone terminal which is connected to the radio system.

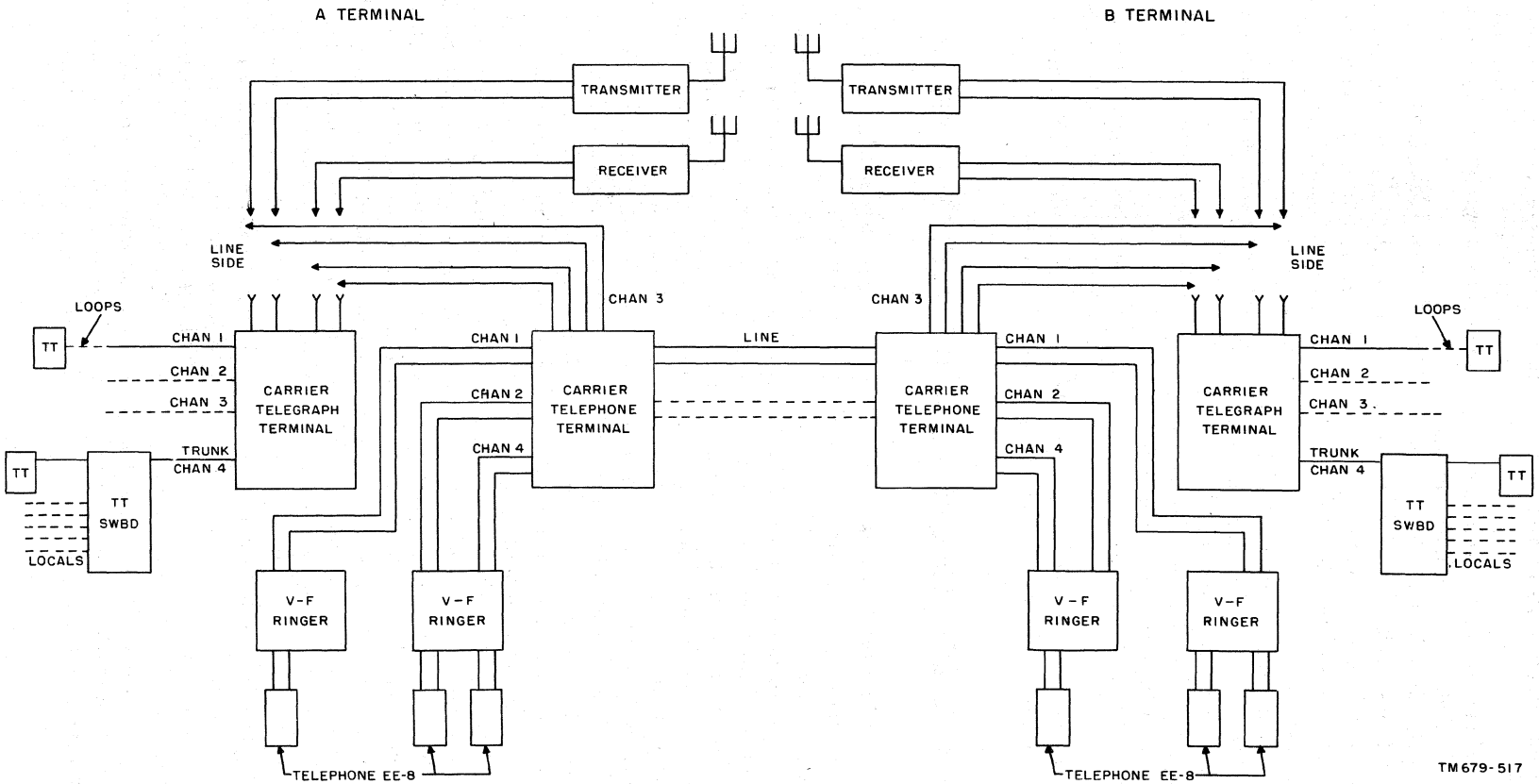


Figure 150. Carrier telegraph system operating over a radio or carrier telephone system.

123. Summary of Introduction to Carrier Telegraphy

a. Telegraphy is a method of transmitting intelligence by means of electrical impulses in accordance with a previously adopted code. Teletypewriters are machines used to transmit and receive these impulses and convert them to printed copy. The impulses transmitted by teletypewriters are d-c impulses.

b. To increase the message capacity of line facilities, the d-c signals are converted to a-c signals, usually in the voice-frequency band, by a carrier telegraph terminal. Each message is transmitted at a different frequency, thus allowing many messages to be transmitted simultaneously over the same line.

c. Since a-c signals are in the voice-frequency band they may be transmitted over carrier telephone systems or radio systems in the same manner as speech signals. If the signals become attenuated they are amplified by a repeater.

124. Review Questions

a. What device is used to transmit and receive d-c signals in telegraph communication?

b. Why are the d-c telegraph signals converted to a-c signals?

c. List the main components in a carrier telegraph system.

d. When is a repeater inserted into a loop circuit?

e. Why are the carrier telegraph signals transmitted over a carrier telephone channel?

f. When would it become necessary to use radio rather than wire?

Section II. TELEGRAPH CIRCUITS AND TRANSMISSION PRINCIPLES

125. Theory of D-C Telegraph Systems

The principles of telegraph transmission is treated briefly in this section.

a. General. Operation of d-c systems of transmission is of two fundamental types: neutral and polar. More complex systems will be discussed later, but they will be basically combinations of neutral and polar operation.

b. Neutral Operation. Neutral or open-and-close operation makes use of a specified flow of current over the line to operate the receiving mechanism to marking or operated position, and zero current over the line for spacing or unoperated position.

c. Polar Operation. Polar mark and space signals are formed by reversing the direction of current flow over the line. When this type of operation is employed, a receiving polar relay is added to the circuit. This polar relay will operate to its mark contact when the current flowing through it is in one direction and will operate to its space contact when the direction of the current flowing through it is reversed. For polar relays see paragraph 128.

d. Transition Point. In either neutral or polar operation, the change from one current condition to the other—that is, from mark to space or from space to mark—is known as a transition.

126. Neutral Half-Duplex Teletypewriter Circuits

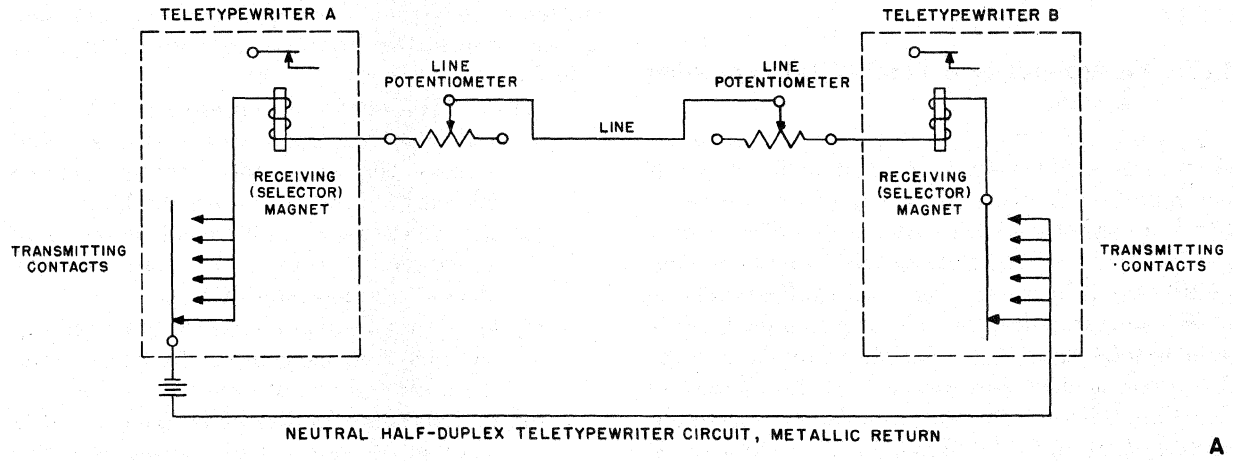
A, figure 151, is a simplified schematic of a neutral teletypewriter circuit employing half-duplex operation. The components of this circuit are a battery, a teletypewriter, a line potentiometer at each end of the circuit, and a pair of wires. When two wires are used to complete the circuit, it is called a *full metallic* circuit. In B, the circuit is completed through ground. This is called a *ground return* circuit. Two of the principal units of the teletypewriter are the transmitting unit and the receiving unit.

a. Transmitting and Receiving Units. The transmitting or sending unit sends out the messages. The message to be transmitted is typed out on a keyboard similar to that of a typewriter. Each key controls a mechanism that produces a group of five current impulses (opens and closures of the circuit). Associated with these five impulses are the stop and start impulses. The 5 pulses are associated with the 5 contacts shown open, and the start and stop impulses are associated with the contact shown closed in A, figure 151. These pulses of current are applied to the receiving magnets at both teletypewriters. The A and B teletypewriter armatures are operated when the current flows through the magnets and

are released when there is no current flowing through the receiving magnets. The movement of each armature sets up a mechanical train of operations for a mark or a space in the receiving units. The receiving magnet is called the *selector magnet*. For each character, the five intelligence impulses are in a unique sequence of marks and spaces. For example, the sequence for a Y is mark-space-mark-space-mark, while for the letter R the sequence is space-mark-space-mark-space. When such a sequence of impulses is received, the character is printed.

b. Circuit. When a transmitter contact is closed at teletypewriter A, current flows from the battery through the transmitting contacts, the receiving magnet, and the line potentiometer at A, through the line, through the line potentiometer, receiving magnet, and transmitting contacts at teletypewriter B, and returns to A through a second wire. Since the current flows through the receiving magnets at both ends of the circuit, the signal is recorded at both teletypewriters. When a transmitter contact opens the circuit, no current will flow in either receiving magnet. Since the signals are formed by intervals of current and no current, they are neutral signals.

c. Half-Duplex Operation. Transmission with this circuit arrangement is restricted to one direction at a time and is called *half-duplex operation*. The transmitting contact at the receiving end must close the circuit when the sending end is transmitting. If the transmitting contact at the receiving end is open (as it is when a space signal is being transmitted) the transmitting end cannot send a



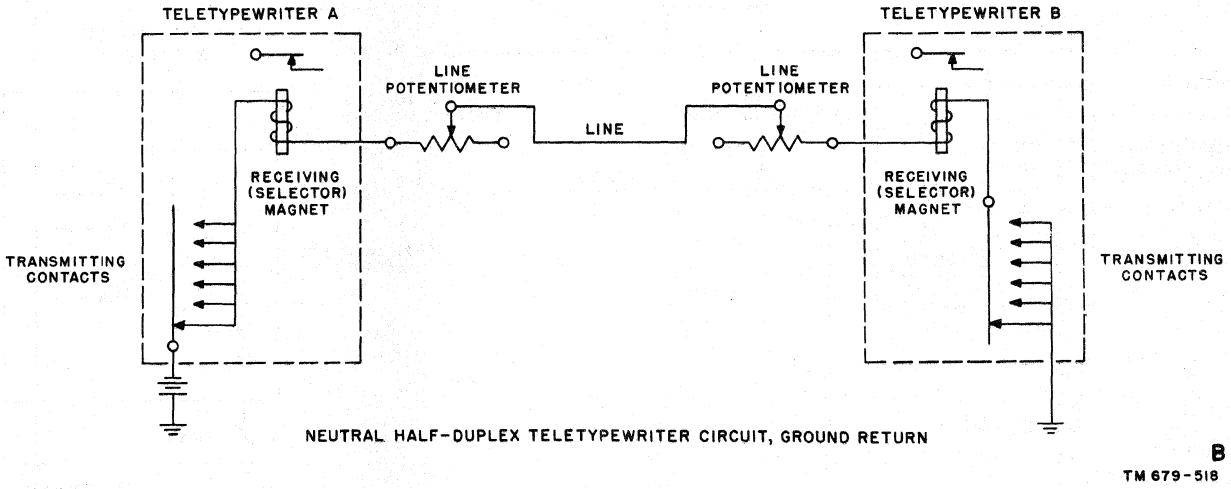


Figure 151. Neutral half-duplex teletypewriter circuits.

mark signal. For this reason, the transmission can be in only one direction at a time. In this arrangement, all of the receiving magnets are in series. The signals transmitted by the A teletypewriter affect the A receiver mechanism so that a copy of the message transmitted is recorded at the same teletypewriter. This is called *home copy*.

d. Full-Duplex Operation. Figure 152 illustrates a circuit employing full-duplex operation and utilizing one teletypewriter at each station. Transmission from A to B is over line 1; transmission from B to A is over line 2. The local receiving magnet is not in series with the local transmitter contacts; therefore, there is no home copy. Because the two circuits are separate, there can be simultaneous transmission in both directions. A circuit which provides simultaneous transmission in both directions is called a *full-duplex circuit*.

127. Waveshapes in a Neutral Teletypewriter System

a. Army start-stop teletypewriter systems operate on a neutral basis at a speed of 60 or 100 words per minute. A word is considered to consist of five letters and a space. At a speed of 60 words per minute, the transmitting shaft makes a complete revolution once every 163 milliseconds to send a start impulse, five selecting impulses, and a stop impulse (fig. 153). The start impulse and the five selecting impulses are each of 22-millisecond duration, whereas the stop impulse is of 31-millisecond duration (or 1.4 times the length of a signal element). The start impulse is always a space impulse which causes the receiving mechanism to begin revolving. The stop impulse is always a mark impulse slightly longer than the other impulses to allow the receiving mechanism to come to a stop before the next combination is received. The time for each character is 7.4 times the length of a single signal element.

b. The theoretical waveshape of the letter Y is shown in figure 153. This waveshape is of a character sent at a speed of 60 words per minute and a line current of 60 ma (milliamperes). Other systems may operate at a speed of 100 words per minute and a line current of 20 ma. When the operator at teletypewriter A (fig. 151) wants to send the character Y to teletypewriter B, the operator at A simply presses the Y key. This causes the transmitting contacts to operate in such a manner that the signal (fig. 153) is transmitted to the line. The sequence of the operation of the transmitter contacts in figure 153 is as follows:

(1) t_0. The operator depresses the Y key; this sets the apparatus in motion and causes the start-stop contact (shown closed in fig. 151) to open the line.

(2) t_1. Twenty-two milliseconds later than t_0, contact 1 closes and causes the line current to increase to 60 ma.

(3) t_2. Forty-four milliseconds later than t_0, contact 1 opens, causing the line to open (contact 2 does not close).

(4) t_3. Sixty-six milliseconds later than t_0, contact 3 closes and causes the line current to increase to 60 ma.

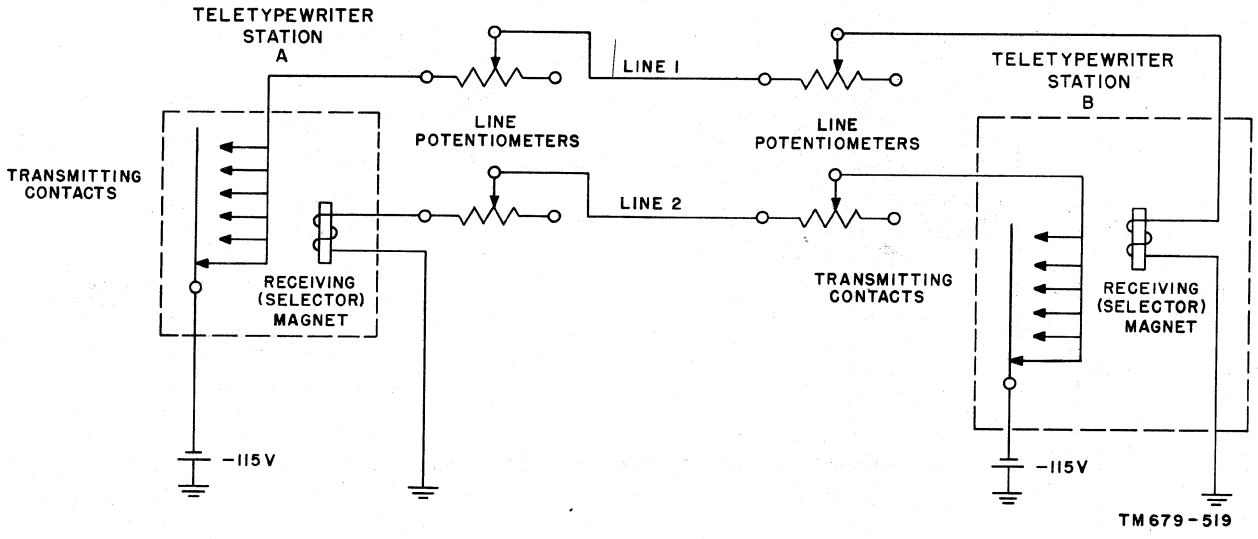


Figure 152. Neutral full-duplex operation.

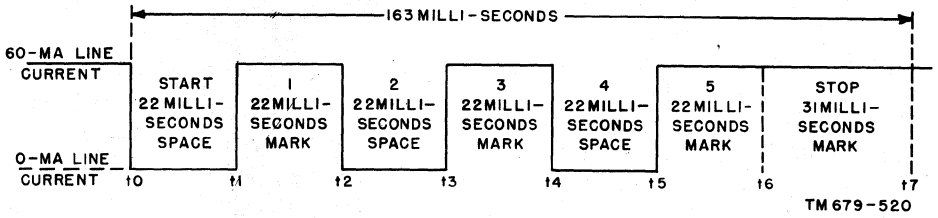


Figure 153. Theoretical waveshape of letter Y.

(5) t_4. Eighty-eight milliseconds later than t_0, contact 3 opens, causing the line to open (contact 4 does not close).

(6) t_5. One hundred and ten milliseconds later than t_0, contact 5 closes and causes the line current to increase to 60 ma.

(7) t_6. One hundred and thirty-two milliseconds later than t_0, contact 5 opens and the start-stop contact closes; the line current remains at 60 ma.

(8) t_7. One hundred and sixty-three milliseconds later than t_0, the sending mechanism is ready for the transmission of the next character.

c. These transmitted signals are perfect signals; that is, the marking and spacing impulses, with the exception of the stop impulse, are equal in length (22 milliseconds), and the mark-to-space and space-to-mark transitions are instantaneous, making the waveshape square. However, characteristics of teletypewriter circuits and apparatus cause the signals to become distorted. A small amount of distortion is normal, but sometimes the signals are so distorted that the receiving unit at the receiving end fails to print the transmitted character.

128. Polar Relays

a. Polar relays are more sensitive receiving units than are selector magnets; they allow a greater operating distance between teletypewriters. Polar relays are the principal components in d-c repeaters and d-c loop circuits in carrier-telegraph terminals.

b. A polar relay with two equivalent windings is shown in figure 154. This relay has two windings of approximately 136 ohms; each winding has the same number of turns and exerts the same magnetizing effect for a given value of current on the relay magnet circuit. Other relays may have three or more windings. The armature is terminated at terminal 1; terminal 4 terminates the mark (M) contact; and terminal 5 terminates the space (S) contact.

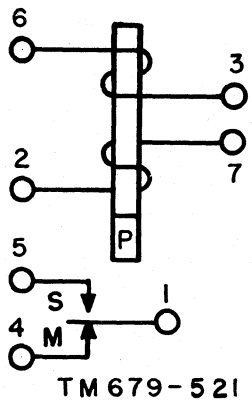


Figure 154. Schematic diagram of a polar relay (255).

c. Since the two windings of the relay illustrated are equivalent, if the current flowing in one winding of the relay is equal to the current flowing in the other, but in the opposite direction, the magnetizing effects will be neutralized and no operation will take place.

d. Basically, a polar relay consists of a permanent horseshoe magnet with the armature of the relay suspended between the pole pieces, and the windings of the relay wound around the armature (fig. 155). In a polar-type circuit, normally only one winding is used; both windings may be employed, however, if connected in a manner that is series aiding. If the relay is properly connected in a circuit to receive polar signals (using the 6–3 winding) and a marking impulse is received, negative potential will appear on the No. 6 side of the relay winding and there will be a positive potential on the No. 3 side of the relay. The flow of current through the winding of the relay establishes a magnetic field. This magnetic field makes the armature of the relay an electromagnet whose north pole is between space and mark contacts. The north pole of the permanent magnet repeals and the south pole attracts the armature,

thus causing the armature to move to the mark contact.

e. When a spacing impulse is received, the potential applied to the No. 6 side of the relay will be positive, and negative will appear on the No. 3 side. This will reverse the conditions stated in *d* above, causing the armature to move to the space contact. In this manner, a polar relay is capable of receiving reversals of current flow from the line.

f. The polar relay may be used in a neutral type of circuit also. If the 3–6 winding is connected in a manner to have the negative potential on the No. 6 side and the positive potential on the No. 3 side during the periods of time when there is a current flow in the circuit (marking pulses), the armature of the relay will move to its mark contact. When there is no current flow in the circuit (spacing pulse) the relay must operate to the space contact. This is accomplished by having the 2–7 winding connected in a local circuit with a negative potential applied to the No. 2 side and a positive potential applied to the No. 7 side. The value of marking current in a neutral circuit is usually adjusted to 60 ma. The current in the 2–7 (bias) winding is adjusted to about 30 ma. When a marking pulse is received and the current builds up to a value above 30 ma, the magnetic effect of the line (3–6) winding becomes greater than the magnetic effect of the bias (2–7) winding and the armature of the relay moves to the mark contact. When a spacing pulse is received and the current flow declines to a value below 30 ma, the magnetic effect of the bias winding becomes greater and the relay armature moves to the space contact.

g. From the foregoing it can also be seen that the arrangement of the winding terminals for a *255 polar relay* is such that if positive potential is applied to a lower numbered terminal, 2 or 3, and negative potential to the corresponding higher numbered terminal, 6 or 7, the armature will operate to the marking or right-hand No. 4 contact as viewed from the front of the relay. If the potentials are reversed, the relay armature operates to the spacing or left-hand No. 5 contact.

h. In circuits where only one battery is applied across the polar relay, the terms positive and negative potential are not strictly accurate. It would be accurate to say that one terminal has a positive potential with respect to the other terminal, and vice versa. For purposes of simplicity, however, the terms positive and negative potential are used throughout this manual.

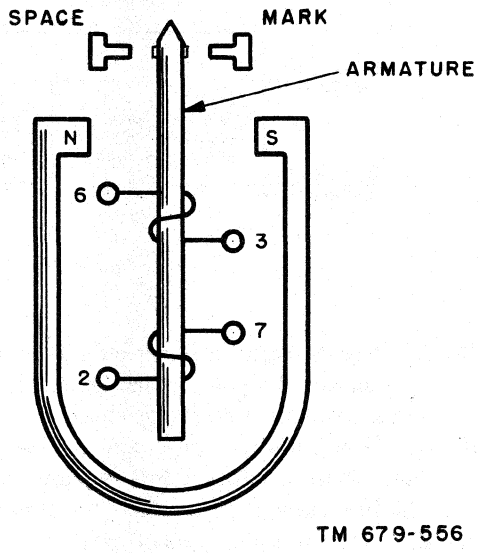


Figure 155. Diagram of 225 polar relay showing permanent horseshoe magnet.

i. The polar relay is sensitive and will operate on a current of approximately 1.2 ma in one winding, or a difference of current of 1.2 ma if the two currents flow through the windings in opposite directions.

129. Polar Operation

a. General. A circuit using one-way polar operation is shown in figure 156. The teletypewriter transmits neutral signals into the A polar relay which relays the signals to the line as polar signals. The B polar relay accepts the polar signals and relays them to the local circuit as neutral signals.

b. Polar Relay Operating on Neutral Signals. When the polar relay is used as a line relay on a neutral circuit, the bias (2–7) winding is locally supplied with 30 ma (adjusted by bias potentiometer) of steady current in a direction to cause the armature to be held to the spacing contact. Positive potential is applied to the high-numbered terminal (7); therefore, the current through the winding is in a direction to cause the relay to operate to the space contact. When a marking impulse is transmitted, the 6–3 winding circuit is closed through the transmitter contacts and positive potential is applied to the low-numbered terminal of the 6–3 winding. Current of 60 ma flows from the negative battery at the transmitter contacts, through the LINE potentiometer, 6–3

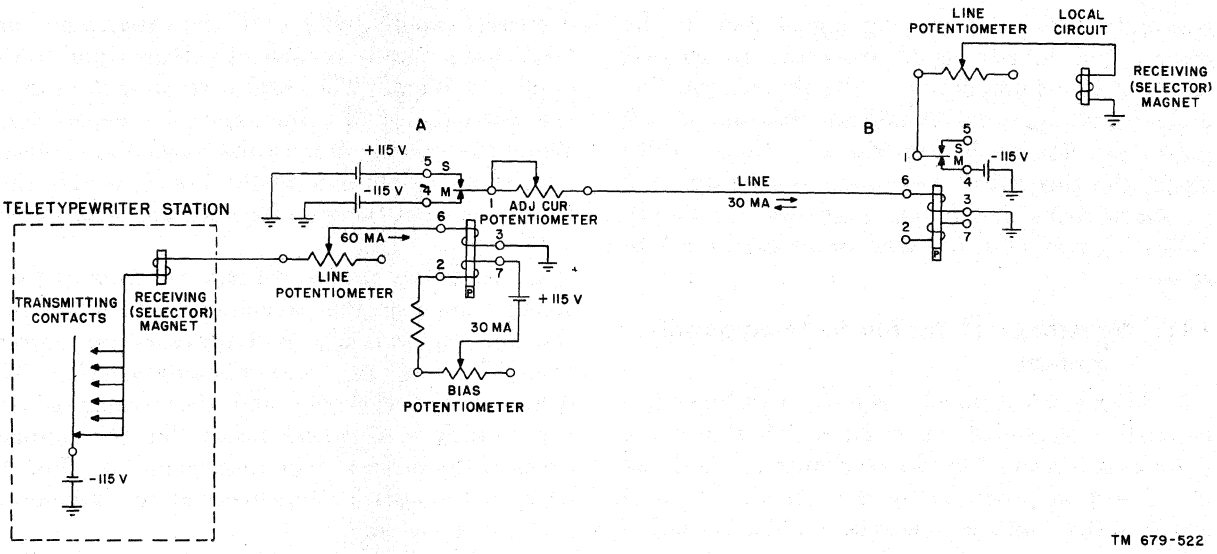


Figure 156. One-way polar operation.

winding of the polar relay to ground, and returns through ground. The magnetic field induced by the 60 ma of line current in the line (6–3) winding overcomes the magnetic field induced by the 30 ma of locally supplied current in the bias winding. This causes the armature to move to the marking (M) contact under the influence of an effective magnetic force resulting from 30 ma of current. When a spacing impulse is sent, the transmitting contacts open the line circuit and no current flows in the 6–3 winding. The armature of the polar relay is moved to the spacing (S) contact by the 30 ma of current in the bias winding.

c. Polar Relay Operating on Polar Signals. When the A polar relay is on the mark contact, current flows from the negative terminal of the battery, through the mark contact, the ADJ CUR potentiometer, the line, the 6–3 winding of the B polar relay to ground and returns through ground. The current is in a direction to operate the B polar relay to the mark contact. When the A polar relay is on the space contact, the positive terminal of a battery is applied to the line. The direction of current flow through the line and through the B polar relay is reversed, causing the relay to operate to the space contact. The polar relay operates on reversals of current flow; therefore, no bias circuit is necessary. The potential of the two batteries applied to the contacts of the A polar relay must be equal so that the magnitude of current for a marking and spacing impulse is equal. The signals in the local circuit are neutral. Current flows through the receiving magnet when the B polar relay is on the mark contact and no current flows in the local circuit when the relay is on the space contact. There is transmission in one direction only with this circuit arrangement.

130. Theoretical Waveshape in Polar Teletypewriter Systems

In polar operation, the sending time for a character is also divided into seven intervals, six of which are 22 milliseconds in duration, and the seventh 31 milliseconds. This is shown in figure 157, which illustrates the theoretical waveshape of the letter Y in polar sending. Marking impulses are transmitted in the first, third, and fifth signal intervals, and, for the stopping signal, by current pulses of 30-ma amplitude. Spacing impulses are

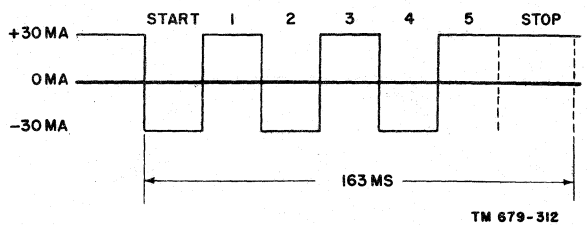


Figure 157. Theoretical waveshape of letter Y in polar operation.

transmitted for the starting signal and in the second and fourth signal intervals by current pulses of 30-ma amplitude. Actually, the positive and negative signs merely indicate that the current pulses are flowing in opposite directions. Once again, the theoretical waveshape is characterized by sharp rising and falling edges for the pulses, indicating instantaneous transitions from mark to space.

131. Waveshape Distortion in Teletypewriter Systems

a. Figure 158 illustrates the effect of line characteristics on signal waveshapes. A shows the pulse as it is formed by the transmitter. B shows the distortion produced by the series inductance which is the result of line relays and retardation coils. The inductance delays the build-up of current on the space-to-mark transition and prevents a sharp decay of current on the mark-to-space transition. The effect of capacitance is shown by the shaded area in C. When a voltage is applied to the line for the transmission of a mark, the voltage must charge the capacitance to ground before it can deliver current to its load (selector magnet). There is instantaneously maximum current flow to the capacitance and minimum to the line. The current to the capacitance decreases expotentially until the capacitance has been charged up to a value of voltage equal to the applied voltage. The total current is the sum of the currents to the line and the capacitance. When the current flow to the capacitance ceases, all the current flow is to the line. At this time the line current takes on its constant, steady-state value.

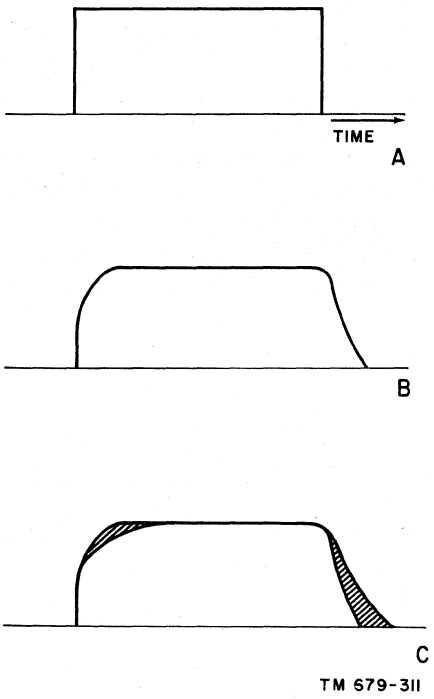


Figure 158. Waveshape distortion in neutral operation.

b. When the applied voltage is removed for a spacing impulse, the capacitance discharges into the line expotentially, and prevents the current from dropping to zero instantaneously. The delay in the charging and discharging of the capacitance is a second reason for the rounded edges of the pulses. The total rounding off of the edges is the sum of the effects of the inductance and the capacitance.

c. In long lines, the current may not reach its steady-state value during the marking interval because of resistance and capacitance of the circuit if the transmitting speed is too great. This is a cause of distortion.

(*1*) Figure 159 illustrates how an increase of signaling speed increases the distortion of a received signal. A, of figure 159, shows a waveshape of a voltage pulse (the diagram at the left) applied during time T at the sending end of the line. On the right is the current waveform of the same pulse at the receiving end of the line. Although the waveshape is distorted, the current not only reaches its steady-state value but holds this value for about one-half of the signal interval.

(2) B, figure 159, shows a voltage pulse formed by applying a voltage during time T/2 at the sending end, and also shows the resultant current waveform at the receiving end of the line. Notice that the current does not reach its steady-state value during the signal interval and, consequently, is distorted more than the signal in A.

(3) C, figure 159, shows that when the time interval of the voltage pulse at the sending end is T/4 (the signaling speed is increased four times that of A) the current builds up to about one-half of the steady-state value.

(4) As the length of the line is increased, the amount of distortion is proportionally increased.

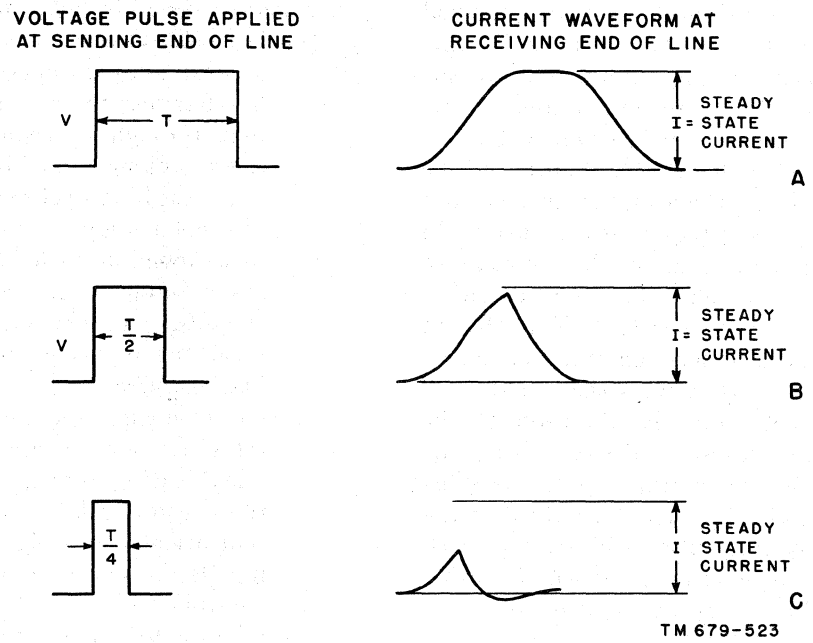


Figure 159. Increase in distortion with increase of signaling speed for a given length of line.

d. Telegraph distortion generally is considered to be divided into three types of components—*bias distortion, characteristic distortion,* and *fortuitous distortion.*

132. Bias Distortion

a. Definitions of Distortion.

(1) The magnitude of distortion is expressed in percent of a unit pulse. Bias, which is the simplest and common component of distortion, may be marking (positive) or spacing (negative).

(2) Marking bias appears as a uniform lengthening of all of the marking pulses, and an equal uniform shortening of all of the spacing pulses.

(3) Spacing bias appears as a uniform shortening of all of the marking pulses, and an equal uniform lengthening of all spacing pulses.

(4) Zero bias is that state in which the marking pulses are equal in length to the spacing pulses.

b. Causes of Bias.

(1) Bias is caused by an improper relation between the levels at which the relay or other receiving device responds and the steady-state marking and spacing levels of the signal. This condition is caused by the presence of inductance and capacitance in the circuit, changes in the value of current, and improper values of operate and release currents of the receiving device. The value of current in a relay which causes the armature to pull up is called the *operating* current of the relay. The value of current which causes the armature to fall back is called the *release* current of the relay. The release current is smaller in value than the operating current.

(2) These values are indicated on the waveshapes in figure 160 as the points O and R. The duration of the signal repeated by the relay is the time, T. In the theoretical waveshape shown in A, figure 160, the operation and the release of the relay take place immediately upon the transmission and completion of the impluse. The operating time, T, is exactly equal to the duration of the pluse. B, C, and D show the waveshapes received over lines of different transmission characteristics, and the effect of waveform distortion on operating time.

(3) In B, points O and R have been delayed because of the time T required for the current to reach the operating and release value in C, the points O and R have been

delayed even more, and by unequal amounts, because an increase in the inductance in the circuit causes the current to build up and decrease more slowly. In D, the points O and R are at greater values because the tension of the spring holding the armature in place has been increased, requiring the current to have a greater value for the armature to pull up. Here, also, there is a time delay before the relay operates. Note that in B, C, and D operating time T, during which the relay repeats the signal, may be different from the duration of the transmitted pulses. Positive bias, the increase of the marking pulse, is illustrated in C. Negative bias, the decrease of the marking pulse, is illustrated in D. It is the unequal delay of the operating time T, that causes bias distortion.

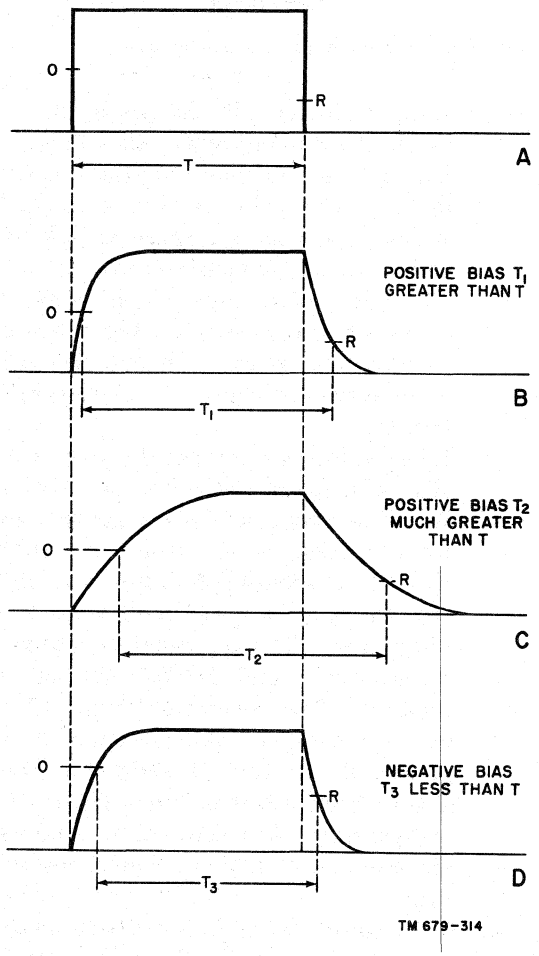


Figure 160. Effect of series inductance on telegraph pulse lengths.

c. Biasing Current in Polar Relays.

(1) The value of line current which will cause a polar relay to operate can be shifted by changing the amount of biasing current through a second winding of the relay. Figure 156 illustrates the effect of a local biasing circuit on the operation of a polar relay. The biasing current is in a direction to hold the relay to the space contact. As a result, the armature cannot move over to the marking contact until the marking current exceeds the biasing current.

(2) The operating and release points of the relay are slightly above and below the relay biasing current. The positions of these points relative to the biasing current are shown in figure 161. By varying the biasing current, it is possible to control the delay of a relay transition relative to the beginning of the corresponding signal transition.

(3) The manner in which the variation of the relay-biasing current can be used to control the bias distortion in a circuit is demonstrated in figure 162. For the maximum value of biasing current, the S-MTD (space-mark) transition delay is greater than the M-STD (mark-space) transition delay. The duration of the marking pulse is about 20 milliseconds, indicating that negative-bias distortion is the result of using too large a biasing current. For the minimum value of biasing current, the M-STD is greater than the S-MTD. The duration of the marking pulse is now about 24 milliseconds, indicating that positive bias distortion is the result of using too small a biasing current. For the intermediate value of biasing current, the S-MTD is equal to the M-STD. The duration of the marking pulse is almost exactly 22 milliseconds, indicating that for some value of the biasing current there is no bias distortion.

133. Advantage of Polar Operation

The waveshape of figure 163 illustrates the major advantage of polar operation. The space-to-mark and the mark-to-mark waveshapes are identical; that is, the space and mark signals, although dis-

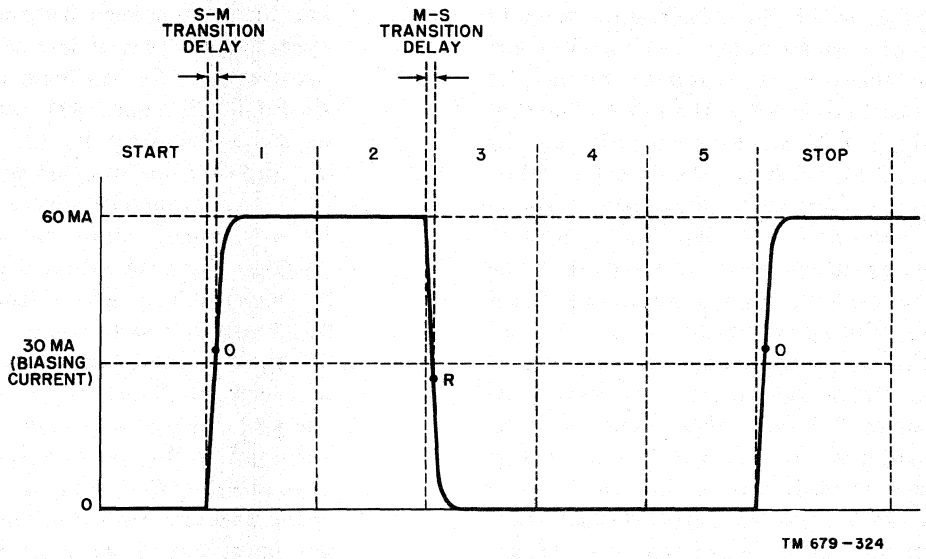


Figure 161. Waveshape of letter A in neutral teletypewriter circuit.

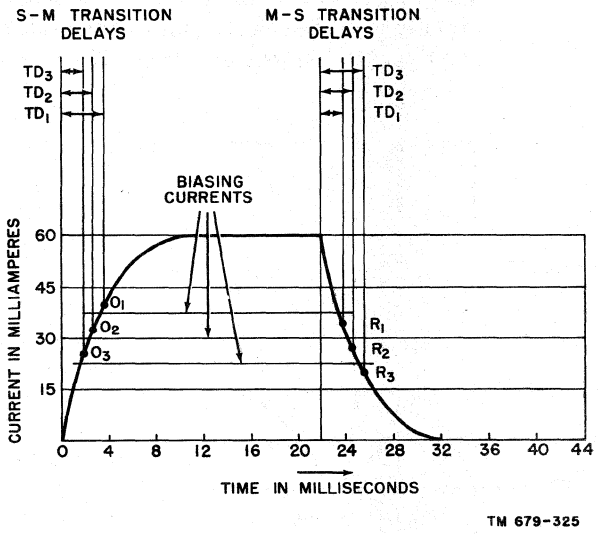


Figure 162. Effect of relay biasing current on signal lengths.

torted, are still of the same length. As a result, the operating points are at the same relative position on each wave. There is no resultant shortening of either the marking or the spacing impulses which could render any character unrecognizable. Thus, there is good operation under adverse line conditions. Unbiased polar transmission depends on three conditions—

a. That equal but opposite potentials be applied at the sending end;

b. That the resistance of the circuit remain constant for both positions of the sending armature;

c. That the operating points of the relay be located symmetrically about the middle of the

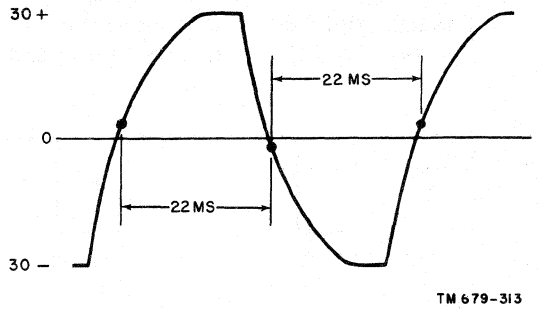


Figure 163. Signal waveshape in polar teletypewriter circuits.

waveshape in order that equal transition delays will be secured.

134. Characteristic Distortion

a. General. Characteristic distortion is best defined in terms of its major properties. First, the distortion takes place when there is a transition, not from a steady-state marking or spacing current as in bias distortion, but from a changing current, because the steady-state value has not yet been reached. Second, it is a distortion of the transmitter pulses that is characteristic of the inductance, capacitance, and resistance of a particular circuit. For some other circuit with different circuit constants, the amount of characteristic distortion is different. Finally, the form of the distortion does not change in sign or magnitude when the marks and spaces are interchanged.

b. Negative Characteristic Distortion.

(1) A, figure 164, shows a theoretically perfect waveshape; B shows the same wave-

shape in which the steady-state value of current is not attained during the short pulse following a long pulse because of the characteristics of the line. The first marking pulse is a long one, allowing the current to reach its steady-state value. The duration of the first space pulse is not long enough to allow the current to decrease to its zero steady-state value before the following space-to-mark transition. During interval 3, the current builds up to approximately the steady-state value before the mark-to-space transition for interval 4. The effect on the short pulse (interval 2) after a long pulse (interval 1) is to shorten it. The time delay from the mark-to-space transition and the point of the relay release at R (shown by D1) is greater than the time delay from the space-to-mark transition and the relay operating at 0 (shown by D2). The short pulse (interval 2) is shortened because the relay operation is delayed less after the short pulse than after the long one. This is shown in B, figure 164, where T1, the duration of short pulse 2, is shorter than T2, the duration of short pulse 3.

(2) The current during pulse 4 decreases to its zero steady-state value before the space-to-mark transition of pulse 5 which is a short pulse. The current again does not reach the steady-state value before the mark-to-space transition of signal 6. The current decreases to approximately the zero steady-state value during interval 6 before the space-to-mark transition of pulse 7. The time delay from the space-to-mark transition of pulse 5 to the operation of the relay at point 0–2 (shown by D4) is greater than the time delay from the mark-to-space transition of pulse 6 to the release point of the relay at point R3 (shown by D5). Thus, the

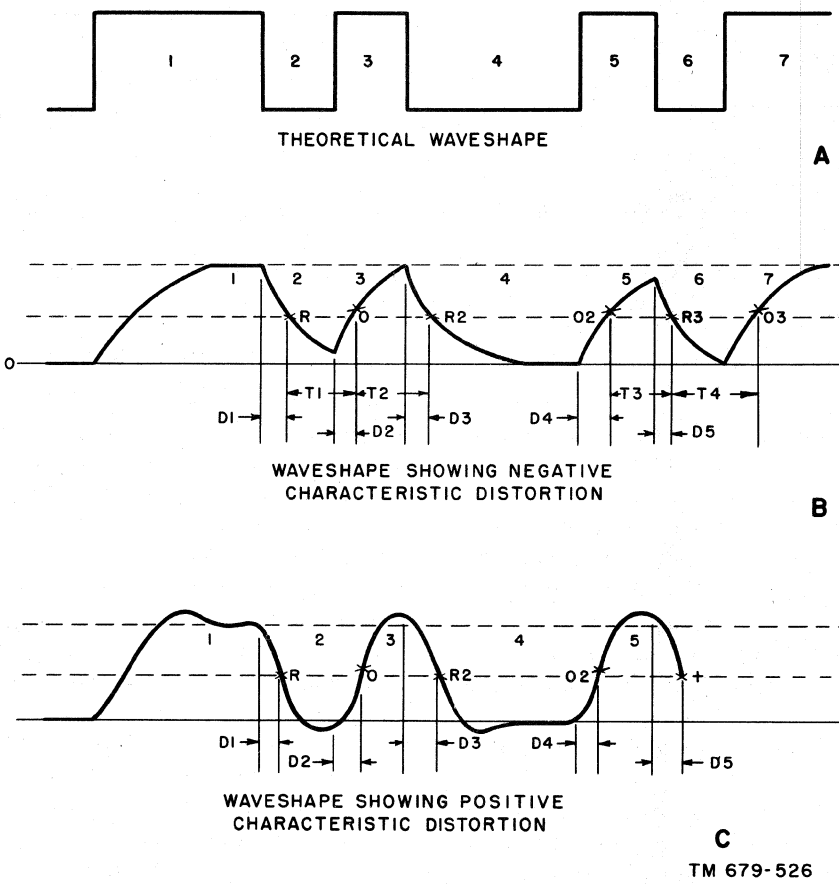


Figure 164. Waveshapes showing characteristic distortion.

short pulse again is shortened. This is apparent from B where T3, the duration of short pulse 5, is shorter than T4, the duration of short pulse 6. *The effect of negative characteristic distortion is to shorten the short pulse after a long pulse,* and this is true whether the pulse is marking or spacing.

c. Positive Characteristic Distortion.

(1) *The effect of positive characteristic distortion is to lengthen the short pulse after a long pulse.* This effect is caused by the operation of the relay being delayed less after a long pulse than after a short one. This is true whether the pulse is marking or spacing. C, figure 164, shows a wave in which the current overswings the steady-state value and fails to return to the steady state within the duration of the shortest pulse. During the first long pulse, the current rises above the steady-state value but decreases to approximately the steady state before the first mark-to-space transition. During short pulse 2, the current decreases below the steady-state value and fails to return to it before the space-to-mark transition of pulse 3. During pulse 3, the current overshoots the steady-state value and does not complete the return to it during the pulse. With this condition, D1 (the time delay between the transition and the relay's release) is less than the time delay of the operation of the relay which is shown by D2. As a result, the duration of the short pulse 2 after the long pulse 1 is increased.

(2) Since the time interval of D4 is less than the time interval of D5, the length of the short marking pulse 5 after the long spacing pulse 4 is also increased. When a short pulse is lengthened after a long pulse, while the pulse is marking or spacing, the effect is known as positive characteristic distortion.

(3) If a wave performs a damped oscillation before settling to a steady state, it is possible that it will produce a negative characteristic effect on certain transitions and a positive characteristic effect on others. Because characteristic distortion is dependent on the previous history of the signal, it is referred to as intersymbol interference.

135. Fortuitous Distortion

In comparison with bias and characteristic distortion, both of which are systematic forms of distortion (that is, they occur in the same way at all times) fortuitous distortion causes the lengths of the received impulses to vary in an erratic manner. The reason for this is simply that the causes of fortuitous distortion do not occur in any regular way. The causes are the chattering or sparking of relays; paralleling telegraph circuits and the "crossfire" introduced by these circuits; natural causes such as lightning; paralleling power lines; intermittent shorts, opens, and grounds on the transmission line.

136. Teletypewriter Margin Measurements

a. The need for measuring distortion, as well as for obtaining intelligible copy even though the signals are distorted, is apparent. Within certain limits, the teletypewriter itself can be used to accomplish this.

b. A relay repeats a signal without time delay in transition. However, if the relay is incorrectly adjusted or incorrectly biased, the length of the marking to spacing pulses may be incorrect. For adjustment of a relay for transmitting unbiased signals, see paragraph 141. Figure 165 shows the sequence of circuit conditions produced by the relay contacts in repeating the letter R (with zero bias) to a receiving mechanism. Shaded areas represent marking intervals, and the unshaded areas represent the spacing intervals. The solid small blocks superimposed upon the received signals represent the selecting interval, or those parts of signals that are used by the selecting mechanism of the receiving machine. The receiving mechanism requires only about 20 percent of that impulse. When the signals are unbiased and the selecting interval is taken from the center of the pulses, the time length from the edge of each selecting interval to the adjacent transition is two-fifths of the length of the impulse. This indicates that the transition may be shifted toward the selecting interval as much as 40 percent before an error is recorded when an ideal signal is received.

c. The receiving unit of the teletypewriter is equipped with a mechanism which causes a latch assembly to be moved mechanically through an arc corresponding to the length of a unit segment

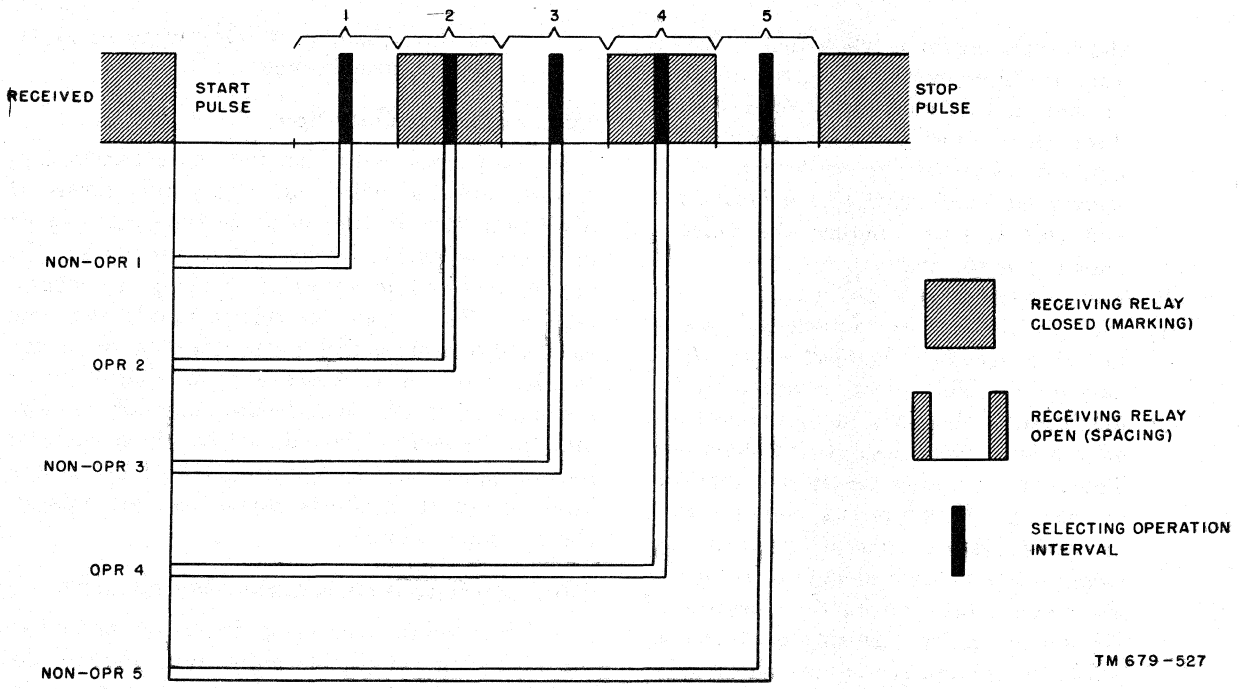


Figure 165. Received signal of character R with zero bias.

or perfect impulse. By this means, all the selecting intervals may be shifted with respect to the beginning of the start impulse over a scale range equal to a perfect impulse (22 milliseconds for 60 words a minute). This mechanism is known as the rangefinder. It is equipped with a scale from 0 to 120 (fig. 166). One hundred divisions on this scale represent an arc equal to a unit segment or 100 percent of an impulse. When the range finder arm is moved toward the lower numbers, it is in effect moving all selecting intervals to the left. When the arm is moved to the higher numbers, the selecting intervals are moved to the right. When an impulse or signal is perfect, the range of the impulse is 100 percent, equaling 0 to 100 on the range finder.

d. Since the ideal position for the selecting intervals is the center of the impulse, the rangefinder arm should be set on 50. The selecting mechanism must use 20 percent of the impulse, and since the rangefinder is set on 50, the 20 percent will come from the exact center of the impulse and leave a total of 80 percent, 40 percent on each side (figs. 165 and 166). Therefore, the range arm can be moved 40 points each way without receiving a wrong impulse, resulting in a range of 10 to 90 (50 minus 40 to 50 plus 40) or 80 points.

e. Figure 167 shows what happens when a marking bias of 40 percent is present. Notice that the selecting intervals may still be moved to the left by 40 percent. The marking impulse becomes longer; this makes it impossible to move the selecting intervals of the spacing impulses to the right without picking up the wrong impulse (mark instead of space). The rangefinder arm, then, can be moved 40 percent to the lower numbers, or on 10 on the rangefinder. If the rangefinder is moved to the higher numbers, however, the selecting interval of the spacing impulse would be at a marking impulse when it should be at a space and thus cause printing of a wrong character. The range in this case would be 10 to 50, or 40 points on the rangefinder.

f. Figure 168 shows the character R received with a 40-percent spacing bias. In this case, if the rangefinder arm were moved to the lower numbers (selecting intervals to the left) a wrong impulse would be picked up (spacing instead of marking) by two of the selecting intervals on the marking impulses. This would cause the wrong character to be printed. However, the arm may be moved to the high numbers (selecting intervals to the right) by 40 points without receiving a wrong impulse. The range in this case would be 50 to 90 on the rangefinder scale.

g. To measure net effect of all kinds of systematic distortion, or position of received signals, the rangefinder arm is moved first in one direction

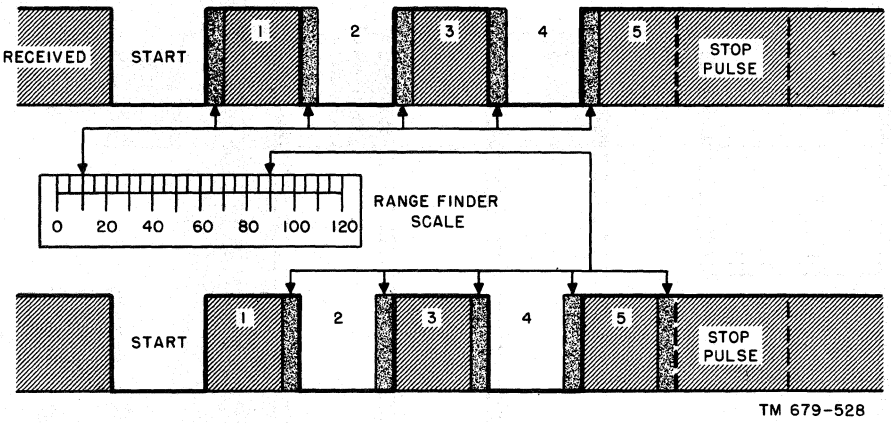


Figure 166. Shifting instants of selection on letter Y.

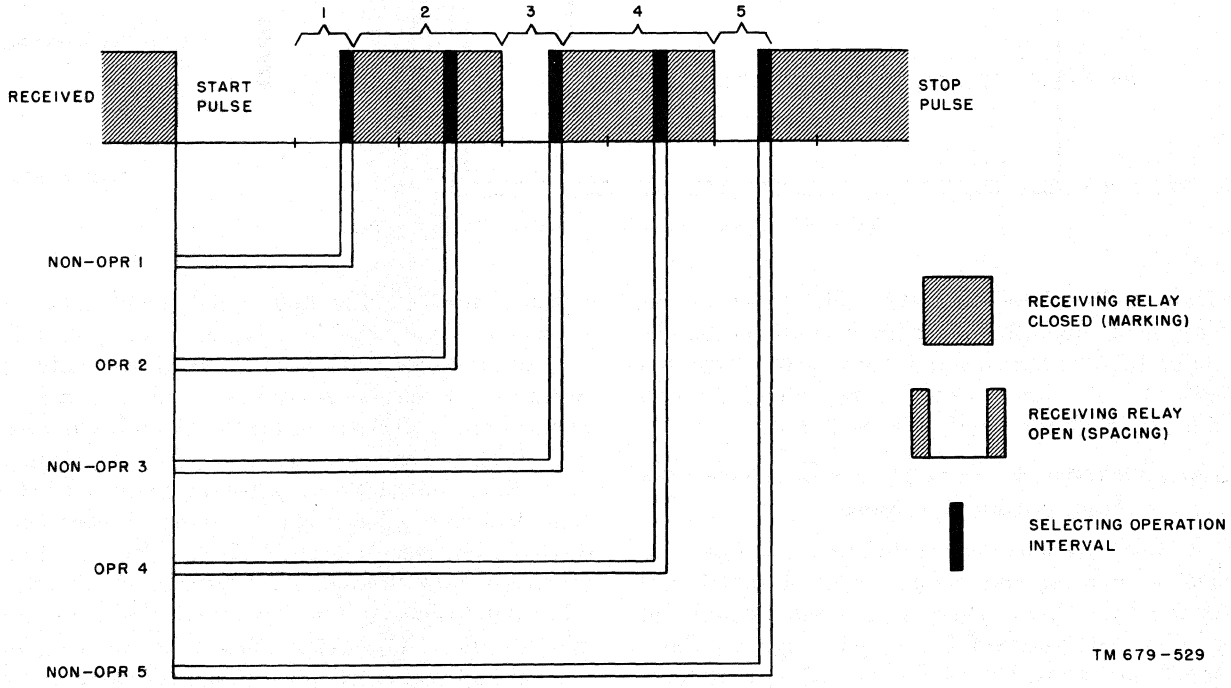


Figure 167. Received signals of character R (marking bias).

until errors appear in the copy and then moved back slowly until these errors are first eliminated. Similarly, the rangefinder arm is moved the maximum distance in the opposite direction. These two scale readings then give the operating margin of the signals under test. On perfect signals, the margin would be from 10 to 90, 80 points, but with either marking or spacing bias this margin would not be so great. With marking bias the high part of the range would be affected, and when a spacing bias is present, the low part would be affected.

h. Margin measurements, in addition to showing the distortion present in the received telegraph signals, also show speed differences between the sending and receiving machines. The effect of a slow sending speed is to cause each unit to be greater than 22 milliseconds and each transition to occur progressively later than it should. The effect on the margin of operation is to raise both limits, the lower limit being raised much more than the upper limit. For example, a margin of 35 to 100 indicates that the sending speed is slow. On the other hand, the effect of a fast sending speed is to cause each unit to be smaller than 22 milliseconds and each transition to occur progres-

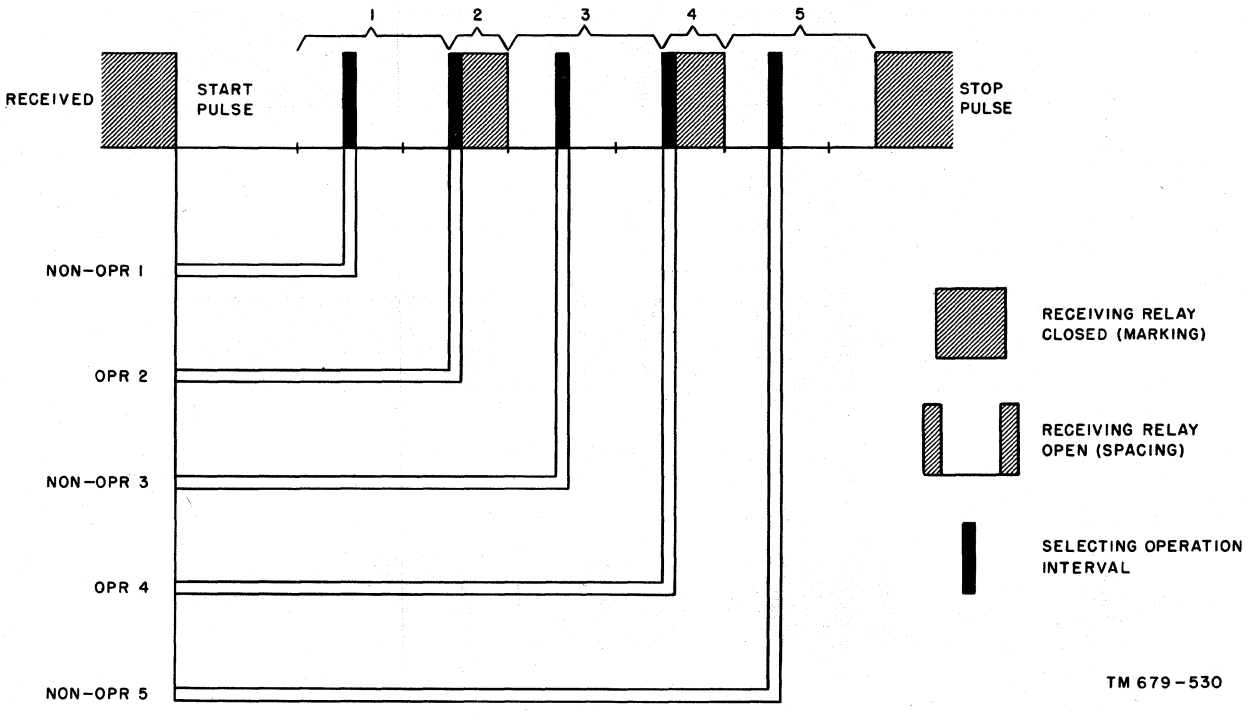


Figure 168. Received signals of character R (spacing bias).

sively earlier than it should. The effect on the margin of operation is to lower both limits, the upper limit being lowered much more than the lower limit. For example, a margin of 5 to 60 indicates that the sending speed is fast.

137. Summary of Telegraph Circuits and Transmission Principles

a. D-c systems fall into two fundamental classes—neutral and polar. Neutral signals are formed by a flow of current for a *mark* signal and a no-current interval for a *space* signal. Polar signals are formed by a reversal of current flow. The change from one current condition to another is known as a transition.

b. With half-duplex operation transmission can be in only one direction at a time. When full-duplex operation is used there may be simultaneous transmission. If two wires are used to complete the teletypewriter circuit, the circuit is called *full metallic;* if one wire and ground are used to complete the circuit, it is called a *ground return* circuit.

c. Army start-stop teletypewriter systems operate on a neutral basis at speeds of either 60 or 100 words per minute. A word is considered to consist of 6 characters, including one for a space at the end of the word. Each character consists of 7.4 signal elements. The first signal (start pulse) is always a space; the last signal (stop pulse) is always a mark which is 1.4 signal elements in duration. Each character has a unique arrangement of marks and spaces for the five other pulses.

d. Polar relays are more sensitive receiving units than the receiving (selector) magnet of the teletypewriter. They allow a greater operating distance between teletypewriters. The direction of current flow through the windings of the polar relay determines to which contact the armature will operate. The polar relay may be used in neutral and polar circuits. When the polar relay is in a neutral circuit, a biasing circuit is used; when it is in a polar circuit, the bias winding of the polar relay usually is disconnected because the polar signals will operate the relay.

e. All d-c signals are subject to distortion. This distortion may be caused by the line facilities, man-made electrical disturbances, crossfire, etc. The types of distortion are classified as *bias distortion, characteristic distortion* and *fortuitous distortion.*

f. Bias distortion may appear as marking bias or spacing bias.

(1) Marking bias appears as a uniform lengthening of the marking pulses and a uniform shortening of the spacing pulses.

(2) Spacing bias appears as a uniform lengthening of the space pulses and a uniform shortening of the mark pulses.

(3) Zero bias is that state in which the marking and spacing pulses are of equal duration.

g. Characteristic distortion appears when the transition takes place before the current has reached its steady state.

(1) The effect of *negative characteristic distortion* is to shorten the short pulse after a long pulse.

(2) The effect of *positive characteristic distortion* is to lengthen the short pulse after a long pulse.

h. Fortuitous distortion occurs at random. It may be caused by chattering relays, crossfire, intermittent shorts, or paralleling power lines.

i. Polar operation often is employed when neutral operation fails because the polar signals are not as susceptible to bias distortion. Since Army teletypewriters transmit and receive neutral signals, a device such as a polar relay or a repeater is used to convert the neutral signals to polar signals and vice versa.

j. The selector mechanism of a teletypewriter requires only 20 percent of a signal for operation. The rangefinder is a latch assembly which determines which 20 percent of the signal the receiving unit selects. On the rangefinder is a scale from 0 to 120. From 0 to 100 represents 100 percent of a signal. When perfect signals are received, the range is from 10 to 90 on the rangefinder. As the distortion in the received signals increases, the range over which clear copy can be received decreases. Thus, the range through which a rangefinder may be turned with correct copy being recorded is a measure of the quality of the circuit.

138. Review Questions

a. Draw a theoretical neutral waveshape for the letter Y. Draw a theoretical polar waveshape for the letter Y.

b. Define a *transition.*

c. Explain the terms full *metallic, ground return, half-duplex operation, full-duplex operation* and *home copy.*

d. How many transmitting contacts are there on the teletypewriter transmitting unit? What is the purpose of the start pulse? The stop pulse?

e. Are neutral signals more liable to bias distortion than polar signals? Why?

f. Why is it possible to disconnect the bias winding on a polar relay when it is to receive polar signals?

g. List three causes of bias distortion.

h. How does the variation of the biasing current in a relay control the amount of bias distortion in the signals that the relay transmits?

i. How the amount of distortion in received signals be measured when only a teletypewriter is available?

j. When the range is 10 to 50, do the characters received have spacing or marking bias?

k. What is the difference between negative and positive characteristic distortion?

Section III. LOOP CIRCUITS OF CARRIER TELEGRAPH SYSTEMS

139. Function of Loop Circuits

a. General. The carrier telegraph loop circuit provides a means for transferring the teletypewriter transmitter's signals to the carrier telegraph terminal to modulate the carrier frequency and, conversely, to relay the demodulated signals from the carrier telegraph terminal to the teletypewriter. The signals in the loop may be d-c or a-c.

b. D-C Loops. Many types of d-c loops are employed in the Army carrier telegraph equipment now in operation. The loop operations are neutral operation to negative battery half-duplex; neutral operation to negative battery full-duplex; neutral operation to positive battery half-duplex; neutral operation to positive battery full-duplex; two path polar operation full-duplex; and polarential operation half-duplex. Neutral operation is probably the most important, since it is employed more often than the other types.

c. Voice Frequency Loops. The trend is toward voice-frequency loop circuits in which a frequency shift type of modulation is used. The mark and space signals are of different frequencies. Typical frequencies used are 1,325 for a mark signal and 1,225 for a space signal. One advantage of the voice-frequency loop is that a telephone switchboard is used instead of a teletype switchboard. This decreases the different types of components required to set up a communication network.

140. Neutral Operation to Negative Battery Half-Duplex

a. Requirements for Proper Operation.

(1) In neutral operation, marking impulses are transmitted by pulses of current. This current can flow from the carrier telegraph terminal to a teletypewriter, or from the terminal to a switchboard, only if a closed circuit exists between the power supply at the terminal and the power supply at the teletypewriter. Such a closed circuit can exist only if all pieces of equipment on the line are in series.

(2) The relative polarity of the power supplies at the terminal and at the teletypewriter must be correct. They must be in series aiding, in order to produce the current for the marking pulse. If one power supply is connected to the loop in the wrong polarity, the two power supplies will oppose each other, and no marking pulse will be produced. Since the carrier telegraph terminal may be connected to different teletypewriters at different times, provisions are usually made to make it possible to change the connections of the terminal power supply to place it in the proper polarity with relation to the power supplied at any teletypewriter or switchboard to which it may be connected. This requirement also exists at a repeater or switchboard which may be connected to various pieces of equipment.

(3) Proper polarity must be observed in connecting a repeater and any equipment which employs a relay. If the relay is connected into the circuit with incorrect polarity, the current will flow through it in the wrong direction. An incoming marking pulse may be repeated by the relay as a spacing pulse, and vice versa.

b. Circuit for Neutral Operation to Negative Battery Half-Duplex.

(1) A typical circuit of neutral operation to negative battery half-duplex is illustrated by figure 169. A single line or loop wire exists between the carrier telegraph terminal and the station teletypewriter. Over this line messages can be transmitted in either direction, but not in both directions simultaneously. At the station the negative side of the battery is applied to the line, and the positive terminal of the battery is connected to ground. The loop circuit designation to *negative battery* refers to the polarity of the *station*-battery terminal connected to the loop. When one of the teletypewriter transmitter contacts is closed, the circuit is completed to the loop. When the transmitter contacts are open for transmitting a space signal, the circuit is open. For receiving (idling) conditions, the circuit is closed by the transmitting contact, as illustrated. The receiving magnet is in series with the transmitting contacts so that the transmitted signals will be accepted by the receiving unit and a home copy of the transmitted message is obtained.

(2) SL is the designation for the *send loop* or *send leg.* The point labeled SL on the schematic represents a binding post on the carrier telegraph terminal to which the send loop wire is connected. The loop current is adjusted by the SL RES potentiometer. Resistor R71 is for the purpose of preventing excessive current flow when the resistance of the SL RES potentiometer is accidentally turned out. The SEND and BK (break) relays are connected in series and operate together. The contacts and armature of the SEND relay are connected to the carrier telegraph oscillator circuit. When the SEND relay is on the mark contact, a mark signal of carrier frequency is transmitted, and when the SEND relay is on the space contact, a space signal is transmitted to the distant carrier telegraph terminal. The L2 coil is in the circuit to prevent the telegraph signals from causing interference in a telephone side circuit when the loop is a simplexed pair of a telephone circuit. The purpose of capacitors C53 and C54 and resistor R61 is to prevent arcing at the contacts of the REC relay when it is transmitting signals to the station teletypewriter.

(3) When a positive potential is applied to the high-numbered terminal of a winding of the polar relay, the current flow through the winding is in a direction to

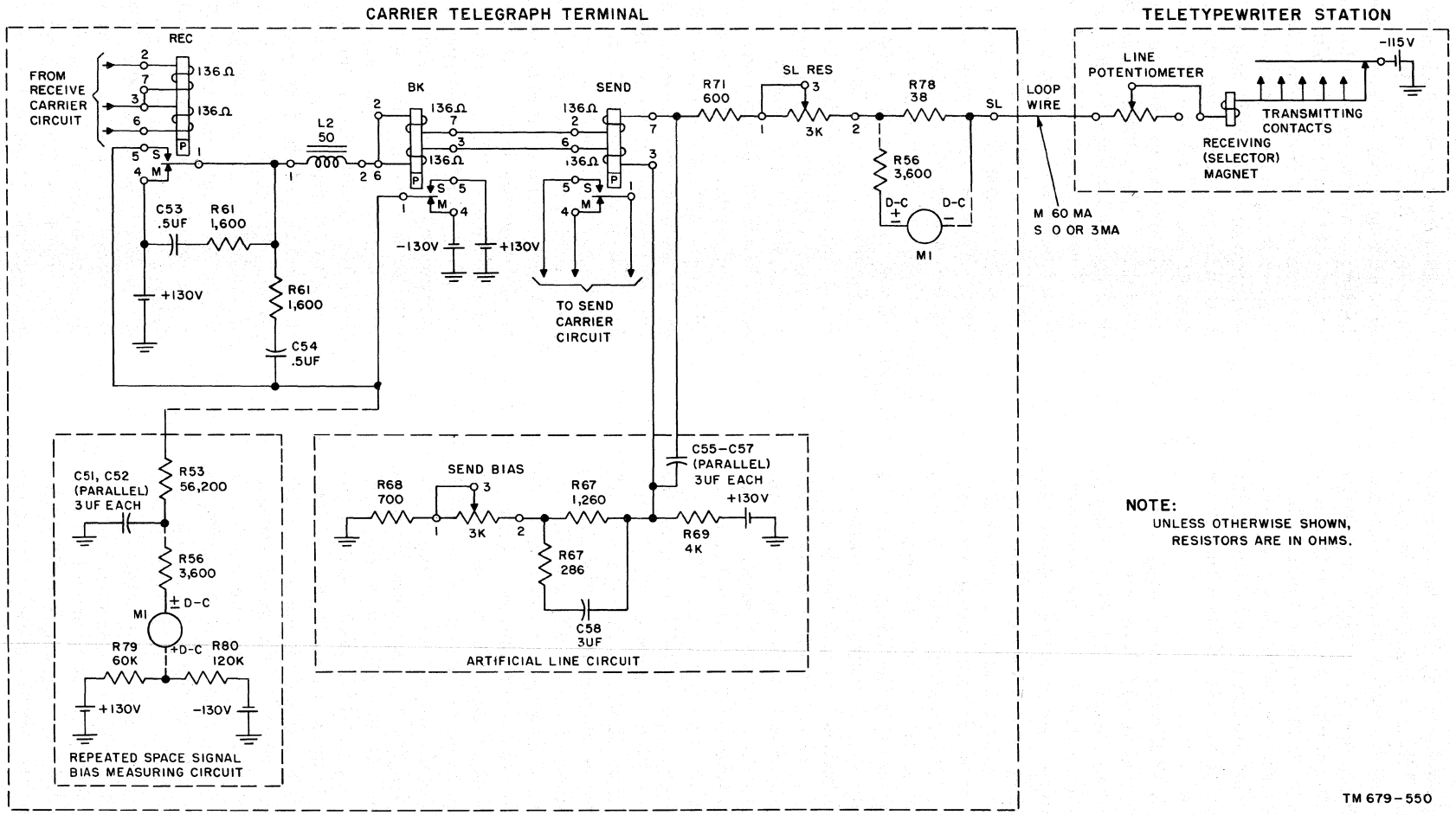


Figure 169. Functional diagram of neutral loop, half-duplex negative battery at station.

cause the relay to operate to the space contact. When a negative potential is applied to the high-numbered terminal of a winding of a polar relay, the current flow is in a direction to cause the relay to operate to the mark contact. However, the winding with the greater current flowing through it will control the operation of the relay if the windings are equivalent.

c. Transmission of Signals From Station Teletypewriter to Carrier Telegraph Terminal.

(1) *Transmitting a mark.* The REC relay at the terminal must be on the mark contact to apply series aiding positive battery to the loop circuit. This means that the distant station teletypewriter must be in an idling (marking) condition and may not transmit copy. A mark signal from the distant telegraph terminal signal keeps the REC relay on the mark contact. A transmitter contact at the local station teletypewriter is closed, allowing a current of 60 ma to flow from the negative battery through the transmitter contact, receiver magnet, loop wire, SL RES, R71, 2–7 windings of the SEND and BK relays, coil L2, to the mark contact of the REC relay and the positive 130-volt battery. The return path in this circuit is through ground. The direction of current flow through the 2–7 winding of the SEND and BK relays is in a direction to cause them to operate to the mark contact. There is also a flow of about 30 ma of current from the positive battery at the REC relay through ground, the artificial line circuit, the 3–6 winding (bias winding) of the SEND and BK relays, the mark contact of the REC relay, and to the positive terminal of the battery. This current flow is in a direction to cause the relays to operate to the space contact. Since the current in the line winding is 60 ma, and that in the bias winding is about 30 ma, the relays will operate to the mark contact. *The SEND relay then permits carrier tone to be transmitted to the distant carrier telegraph terminal.* Notice that there is a flow of current through the receiving magnet of the teletypewriter so that the proper mechanical operation for a mark signal for home copy is set up in the receiving unit.

(2) *Transmitting a space signal.* The transmitting contacts are open for the transmission of a space signal. This opens the loop circuit and no current flows through the receiving magnet or through the 2–7 windings of the SEND and BK relays. Current does flow through the bias windings of the SEND and BK relays in a direction to operate them to their space contacts. *The SEND relay then prevents carrier tone from being transmitted to the distant carrier telegraph terminal.* Since the receiving magnet is not energized, the proper mechanical operation for a space signal is set up in the receiver unit. Thus, a home copy of the transmitted message is recorded.

d. Transmission of Signals From the Carrier Telegraph Terminal to the Station Teletypewriter.

(1) *Transmitting a mark signal.* The teletypewriter at the station teletypewriter must be in the idling (marking) condition to complete the loop circuit. A marking signal from the distant carrier telegraph terminal operates the receive relay to the mark contact. A current flow of 60 ma flows from the negative battery at the station teletypewriter through the loop, the line windings (2 and 7) of the SEND and BK relays, to the mark contact of the REC relay, to positive 130-volt battery and returns through ground to the negative battery at the station teletypewriter. The receiving magnet of the teletypewriter is energized by the current flow and the mechanical operation for a mark signal is set up in the receiving unit. At the same time, a current of about 30 ma flows from the negative terminal of the battery at the mark contact of the REC relay through ground, the artificial line winding, the bias winding of the SEND and BK relays, the mark contact of the REC relay, to the positive terminal of the battery. The 60 ma of current flowing in the line winding of the SEND and BK relays are in a direction to cause them to operate to the mark contact, while the 30 ma, in the bias windings are in a direction to cause the relays to op-

erate to the space contact; therefore, the relays operate to the mark contact. The SEND relay must remain on the mark contact to allow carrier tone to be transmitted to the distant carrier telegraph terminal to hold the REC relay there on the mark contact. *During this time a mark signal is recorded in the receiving unit.*

(2) *Transmitting a space signal.* When a space signal is received from the distant carrier telegraph terminal, the receive relay is operated to the space contact. The 130-volt negative battery at the mark contact of the break relay is applied to the loop by the space contact of the REC relay. The 115-volt battery at the station end of the loop is also negative and connected in series bucking. The total potential in the circuit is now only 15 volts, and the loop current decreases to approximately 3 milliamperes. Since this current is insufficient to operate the receiving magnet of the teletypewriter, *a space signal is recorded in the receiving unit.* Although there is no current flow through the line winding of the SEND and BK relays, there is a current flow from the negative battery at the mark contact of the BK relay through the space contact of the REC relay, the bias winding of the SEND and BK relays to the point where R69 and R67 are connected. Here the current divides; a portion flows through R69 to the positive battery and returns through ground. The other path is through R67, the SEND BIAS potentiometer, R68, to ground and returns through ground. The current flow through the bias winding of the SEND and BK relays is in a direction to cause them to remain operated to mark contact. The SEND relay remains on the mark contact to allow carrier tone to be transmitted to the distant carrier telegraph terminal to hold the REC relay there on mark. The artificial line circuit has an impedance which is about equal to the impedance of the loop so that the signals in the loop will be duplicated in the bias winding. This is to prevent the SEND and BK relays from being falsely operated when the receive relay is transmitting

e. Breaking. When the receiving operator wishes to gain control of the circuit during the reception of signals, he can do so by opening the loop circuit. This is called *breaking.* When the loop is opened and the REC relay is on the mark contact, the SEND and BK relays are operated to their space contacts by the bias current flow in the artificial line circuit. The BK relay space contact then allows positive battery to be applied to the space contact of the REC relay. The polarity of both batteries applied to the contacts of the REC relay is now positive. The current flowing in the bias windings of the BK and SEND relays is in a direction to cause the relays to operate to the space contacts regardless of the contact to which the REC relay is operated. The SEND relay thus continues to send a spacing (breaking) signal to the distant carrier-telegraph terminal as long as the loop is open. This causes the teletypewriter at the distant terminal to receive a steady space. A garbled (incorrect) home copy at the distant station, which is transmitting, tells the operator there that the receifing station is breaking and wishes to transmit. A prolonged breaking signal will cause the distant receiving unit of the teletypewriter to *run open.* This produces a distinctive pounding sound.

f. Neutral Half-Duplex Operation Using a Full Metallic Loop.

(1) When a second wire instead of ground is used for the return path in the loop, the circuit is called *full metallic.* A full metallic loop is used when (a) a *good* ground is not obtainable, (b) there is no rectifier or line unit at the outlying station to supply and apply the battery to the loop at the teletypewriter, (c) the leakage is too high on a ground return circuit, and (d) there is *crossfire,* which is defined as foreign current impulses produced in teletypewriter circuits by the operation of other parallel teletypewriter circuits. Crossfire is caused by a high resistance to ground which couples the telegraph circuits.

(2) Figure 170 illustrates a typical full-metallic loop circuit. Notice that both batteries are supplied at the carrier terminal. The operation is the same as for the ground return half-duplex loop.

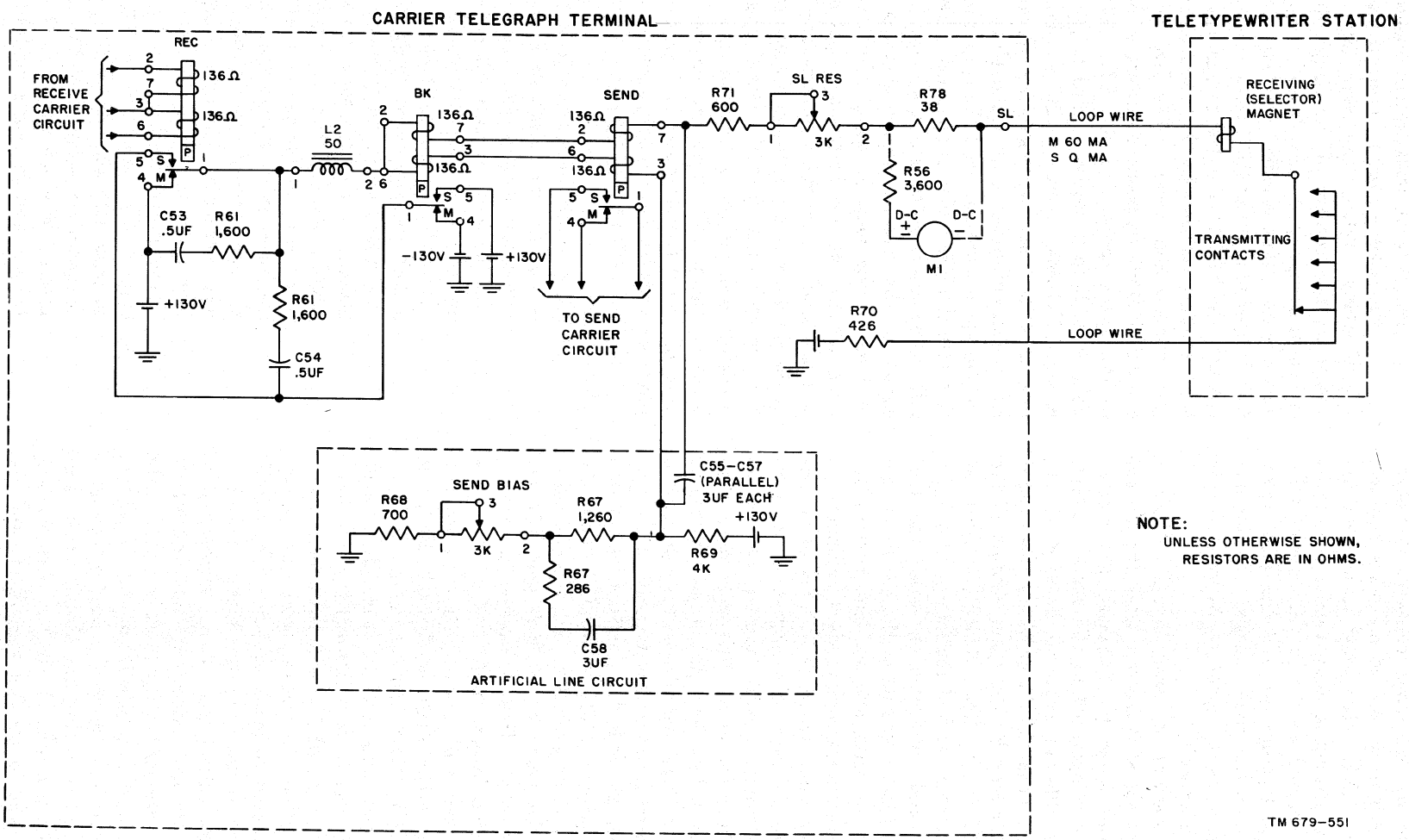


Figure 170. Functional diagram of neutral loop, half duplex, no battery at station, negative terminating battery furnished at carrier telegraph terminal.

The bias measuring circuit has been omitted.

141. Bias Adjustment

The bias condition of signals is determined by the equipment, line facilities, overall length, etc., of the circuit. Since these components are different for different circuits, a provision must be made for the adjustment of current flow in the bias winding of the send and break relays.

a. The manner in which the variation of the relay-biasing current can be used to control the bias distortion in a circuit is demonstrated in paragraph 132*c*. The bias current is adjusted by means of the SEND BIAS potentiometer (fig. 171). R69, R67, SEND BIAS potentiometer, and R68 form a voltage divider across the 130-volt artificial line battery. Point A is therefore at a positive potential, considerable less than the 130-volt source. The positive potential can be varied by adjusting the SEND BIAS potentiometer. This positive potential or voltage is connected in series opposing to the lower 130-volt battery. Variation of the SEND BIAS potentiometer thus varies the potential differences between A and B and this in turn determines the amount of current flow through the SEND and BK relay windings.

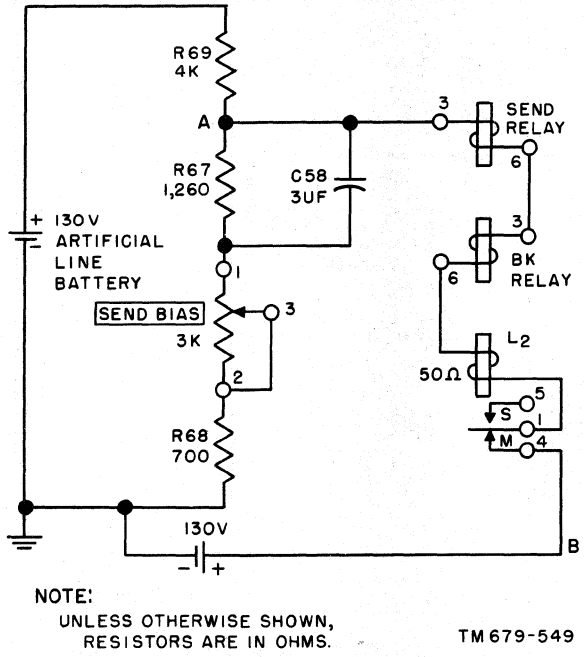


Figure 171. Simplified bias circuit.

b. The purpose of the bias measuring circuit illustrated in figure 169 is to give a *zero* indication when the biasing current in the send and break relays is at the value that will allow the SEND relay to transmit undistorted signals to the distant carrier telegraph terminal. To adjust the biasing current, repeated space signals are transmitted from the teletypewriter to the carrier telegraph terminal. The code for a repeated space signal is illustrated in figure 172. The duration of the spaces is about twice that of the marks. The BK relay operates alternately from mark to space. When the BK relay is on the mark contact, the current flows from the negative terminal of the battery at the contact through the meter to positive battery at R79. When the BK relay is on the space contact, the current flow is from the negative battery at R80 through the meter to positive battery at the space contact of the BK relay. Thus, the current flow through the meter is reversed, causing the needle to oscillate. The length of time that the armature of BK relay is on the contacts for the repeated space signals is determined by the amount of current flow in the bias winding of the BK relay. The current in the bias winding is adjusted by the SEND BIAS potentiometer until the meter needle oscillates about zero. The constants of the bias measuring circuit are such that the meter reads zero when the BK relay is repeating the repeated space signals without bias. Since the SEND relay is in the same circuit as the BK relay, the SEND relay will also transmit practically unbiased signals to the line. In many carrier systems a series of alternate mark and space signals must be received to adjust for zero bias. These may be obtained from a tape with alternate R and Y signals or from dots from a Test Set I–193–C.

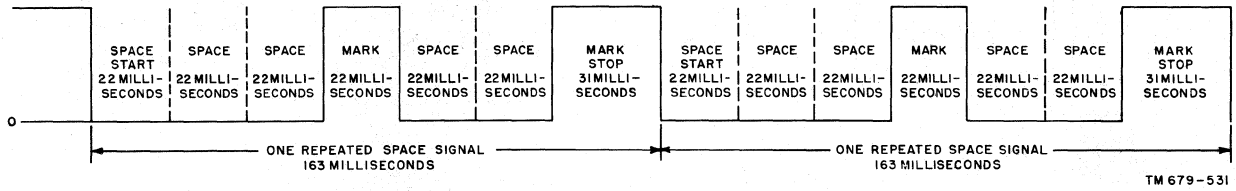


Figure 172. Waveshape of Repeated Spaces.

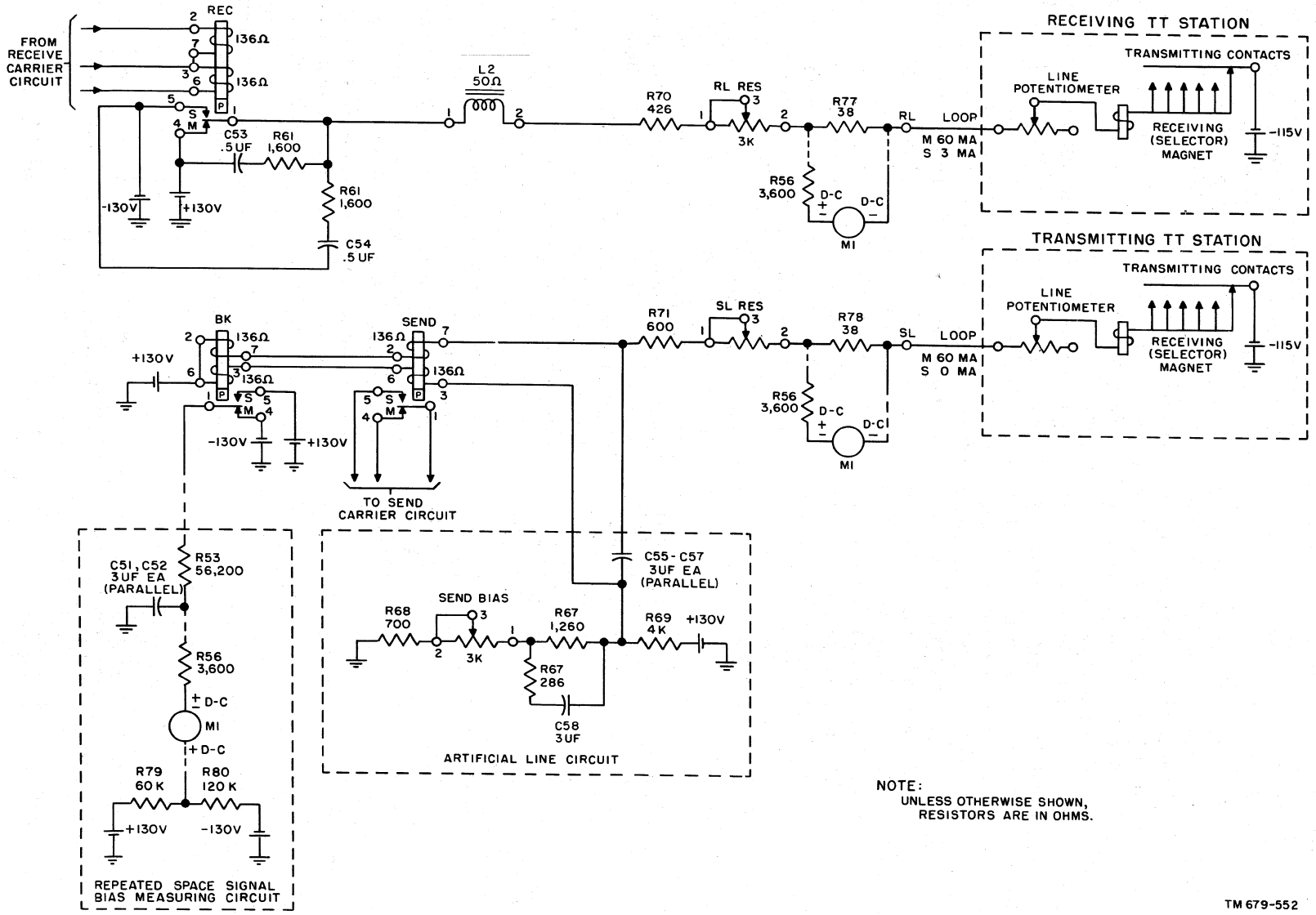


Figure 173. Functional diagram of neutral loop, full-duplex, negative battery at station.

142. Neutral Operation to Negative Battery Full-Duplex

a. General.

(1) Full-duplex neutral operation is illustrated by the circuit shown in figure 173. The channel terminal is arranged for full-duplex transmission; that is, messages can be transmitted in both directions simultaneously. The carrier telegraph terminal is connected to the station by two wires, or simplexed lines, each terminated in negative polarity. The diagram shows two teletypewriters connected to the circuit; one to the receive loop (above) and a second to the send loop (below). The receiving magnet of the teletypewriter connected to the send loop is in series with the transmitting contacts. The receiving magnet will respond to the transmitted signals; therefore, home copy will be received when the teletypewriter transmits. Since the transmitting unit and the receiving unit of the teletypewriter are separate units, it is possible to use only one teletypewriter by connecting the transmitting unit to the send loop and the receiving unit to the receive loop. With this arrangement there will be no home copy of the transmitted message because the receiving magnet is no longer in series with the transmitting unit.

(2) The SEND and BK relays are not in series with the armature of the REC relay for this arrangement. Instead, the SL loop terminates in positive battery connected to the junction of the bias and line windings of the SEND and BK relays. The operation of the REC relay, then, will not affect the operation of the SEND and BK relays. The current in the send and receive loop circuits is adjusted to 60 ma by the SL RES potentiometer and the RL RES potentiometer respectively. Since this is neutral operation, the current in the bias windings of the SEND and BK relays is adjusted by the SEND BIAS potentiometer while repeated spaces are being transmitted to the carrier telegraph terminal from the station teletypewriter transmitting unit connected to the SL loop.

b. Transmission of Signals From Station to Carrier-telegraph Terminal.

(1) *Transmitting a mark.* When one contact of the transmitter unit closes the SL loop circuit, 60 ma of current flows from the negative terminal of the battery at the station through the transmitter contact, the loop wire, R78, SL RES potentiometer, R71, line windings of the SEND and BK relays to positive battery and returns through ground. This current flow is in a direction to cause the SEND and BK relays to operate to the mark contact. At the same time, there is a flow of current, from the negative terminal of the battery connected at the junction of the windings of the SEND and BK relays, through ground, R68, SEND BIAS potentiometer, R67, bias windings of the SEND and BK relays to the positive terminal of the battery. This bias current of about 30 ma is in a direction to cause the relays to operate to space. The SEND and BK relays are operated to mark because of the 60 ma in the line windings. The SEND relay on the mark contact allows the marking carrier tone to be transmitted.

(2) *Transmitting a space signal.* The loop is opened by the transmitter contacts; no current flows in the loop circuit. A current flows in the bias windings, as in (1) above, operating the SEND and BK relays to space. When the SEND relay is operated to the space contact, a space signal (absence of carrier tone) is transmitted by the carrier telegraph terminal. In this circuit, home copy of the transmitted signals is obtained because the receiving magnet of the transmitting teletypewriter is in series with the loop.

c. Transmission of Signals From Carrier Telegraph Terminal to Station.

(1) *Transmitting a mark.* When a marking signal is received from the distant carrier terminal, the REC relay is operated to the mark contact. The positive terminal of the battery at the mark contact of the REC relay is connected to the loop, and current flows in the receive loop because series aiding battery is connected at the station. Current flows through the re-

ceiving magnet of the teletypewriter and a mark signal is recorded.

(2) *Transmitting a space.* When a space signal is received from the distant carrier terminal, the REC relay is operated to the space contact and applies a battery of negative polarity to the loop. The polarity of the battery at the station is series opposing. The 15-volt potential difference does not cause sufficient current to flow in the RL loop to operate the receiving magnet, and a space signal is recorded in the receiving unit. Breaking is not possible with this arrangement because the sending and receiving units are in separate circuits.

143. Neutral Operation to Positive Battery

a. Circuit for Neutral Operation to Positive Battery Half-Duplex.

(1) Should the carrier telegraph terminal be connected to a loop terminal at which the positive terminal of the power supply is applied to the loop, some circuit changes are necessary in order to place the two power supplies (at the station and at the carrier terminal) in series aiding for marking pulses.

(2) The circuit changes are illustrated in figure 174. The polarities of all power supplies are reversed as compared with those in the circuit of figure 169. As a result, current directions are likewise opposite to those of the previous circuit. Further, the 2–7 winding of the SEND and BK relays have been interchanged with the 3–6 winding. If these windings are not interchanged, the current through the loop for a marking pulse would operate the SEND and BK relays to the space contact and false signals would be transmitted to the distant carrier terminal.

(3) In all other respects, the operation of this loop arrangement is similar to that described in paragraph 140.

b. Circuit for Neutral Operation to Positive Battery Full-Duplex. Full-duplex operation to positive battery (fig. 175) is identical with that for neutral operation to negative battery full-duplex, with the circuit changes described in paragraph 142*a*.

144. Two-Path Polar (Full-Duplex) Operation

a. General.

(1) Polar operation, rather than a neutral operation, is employed over long d-c loop sections because the waveshapes of mark-space and space-mark are identical (par. 133). In figure 176, A illustrates a d-c circuit using two-path polar operation employing two d-c repeaters. In two-path polar operation, transmission may be effected in two directions simultaneously. The neutral signals from the A_1 teletypewriter are converted to polar signals by the A repeater, transmitted over line A to the B repeater which changes the polar signals to neutral ones for reception by the B_1 teletypewriter. Transmission from the B_2 teletypewriter to the A_2 teletypewriter is over the second loop wire.

(2) B (fig. 176) illustrates a carrier channel using two-path polar loop operation. On one line, neutral signals are transmitted from the A_1 teletypewirter to the A repeater. The A repeater converts the neutral signals to polar signals and transmits them to the A carrier telegraph terminal, which converts the polar signals to carrier frequency signals and transmits them to the B carrier telegraph terminal. At the B carrier terminal, the carrier-frequency signals are converted to polar signals and transmitted to the B d-c repeater, which converts the polar signals to neutral ones and transmits them to the receiving teletypewriter B_1. Signals can be transmitted simultaneously from the B_2 teletypewriter to the A_2 teletypewriter over the other loop wire. The sequence of operations is the same as in the A_1 to B_1 direction.

b. Circuit for Two-Path Polar Operation.

(1) Figure 177 illustrates a loop circuit employing two-path polar operation. The SEND and BK relays are in a circuit that is not in series with the REC relay. This arrangement is similar to the neutral full-duplex circuits. The bias windings 2–7 of the SEND and BK relays in the carrier telegraph terminal, as well as the bias winding of the R relay in the repeater, have been disconnected. The

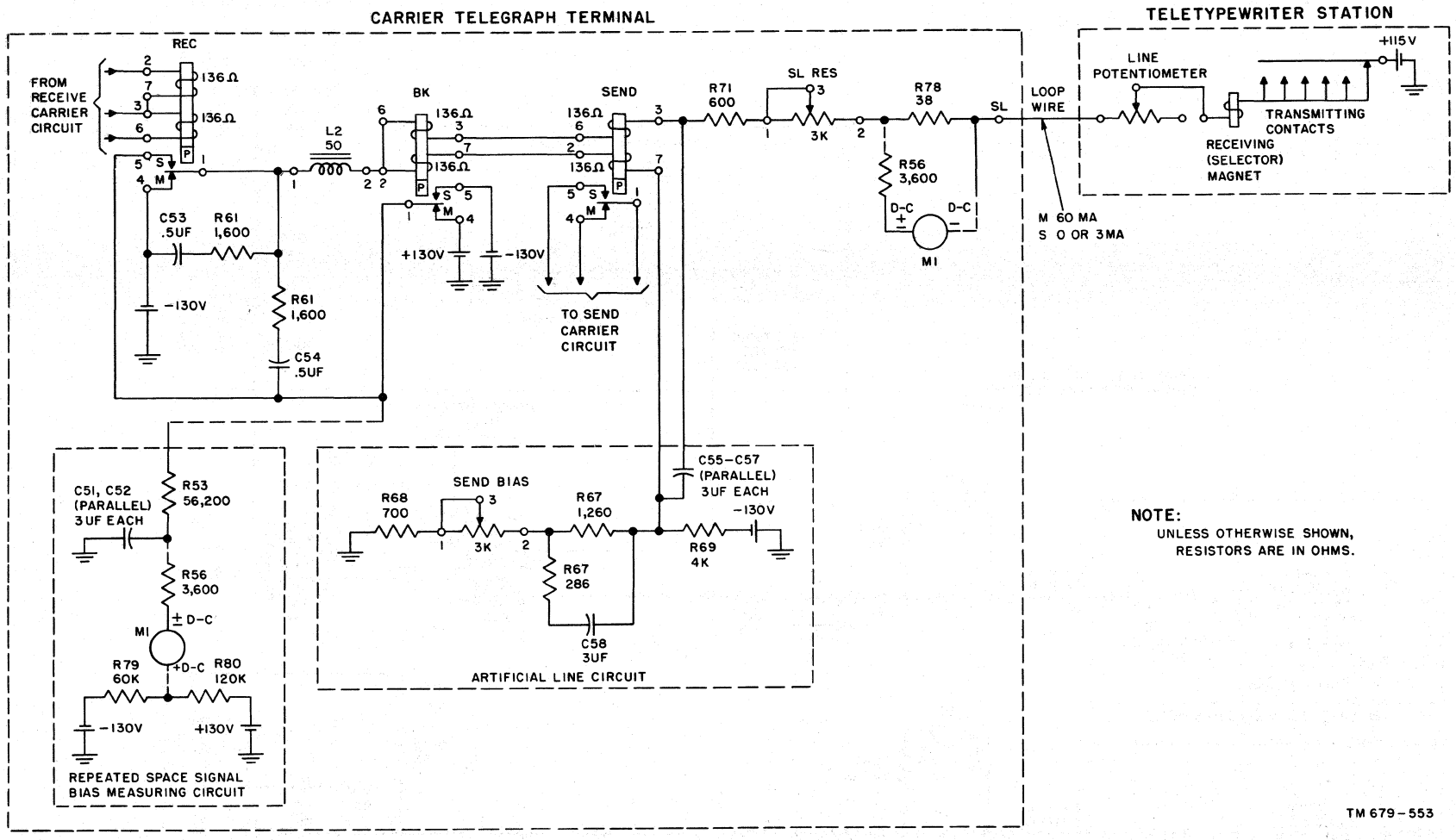


Figure 174. Functional diagram of neutral loop, half-duplex, positive battery at station.

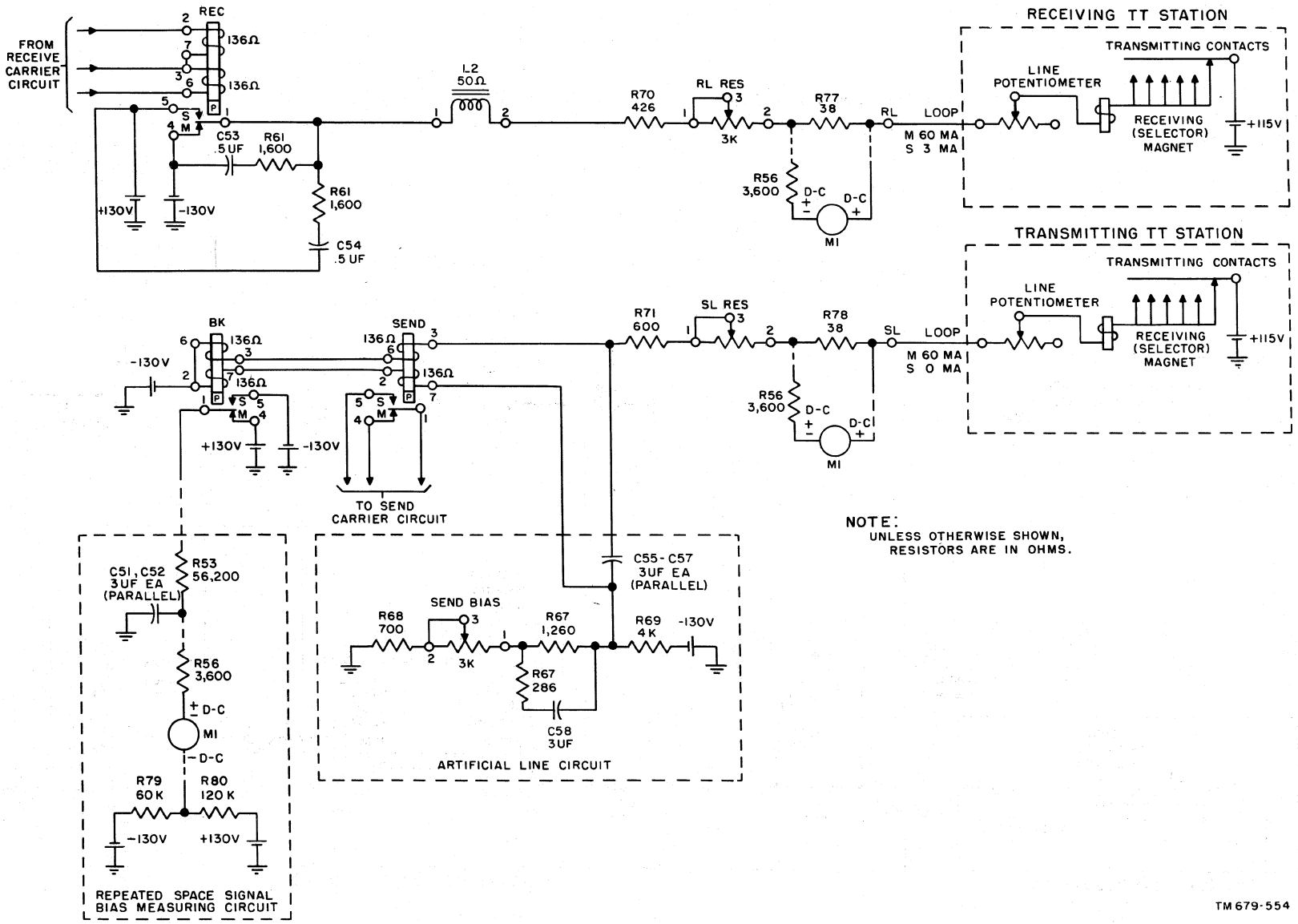
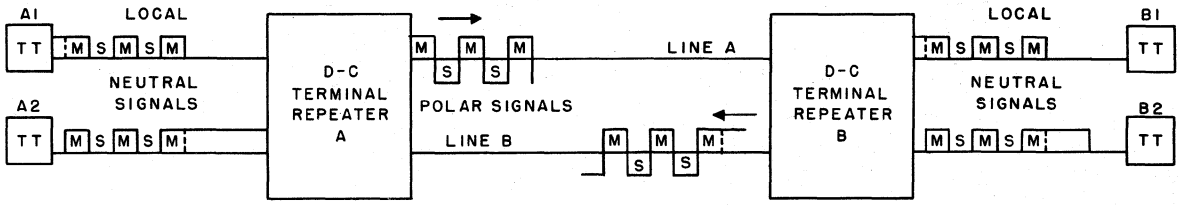
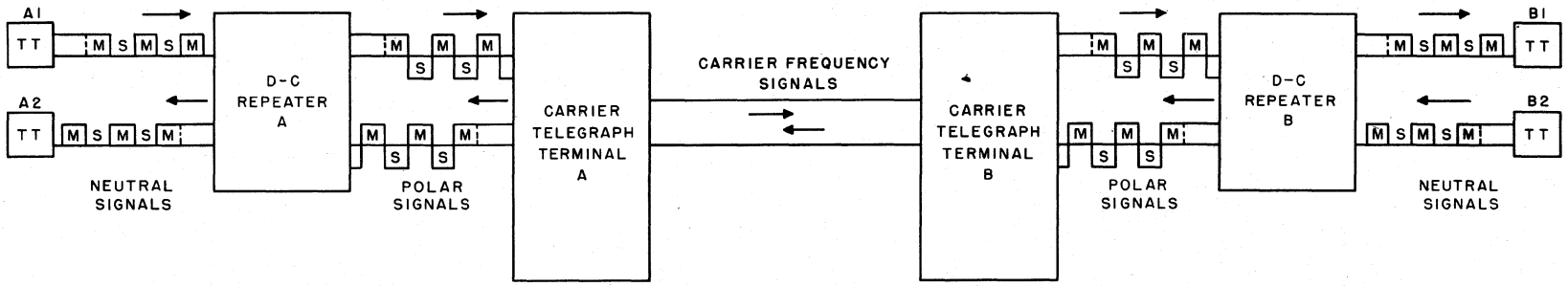


Figure 175. Functional diagram of neutral loop, full-duplex, positive battery at station.



D-C REPEATERS EMPLOYING
TWO-PATH POLAR OPERATION

A



CARRIER TELEGRAPH TERMINALS
OPERATING TWO PATH POLAR (FULL-DUPLEX)

B

TM 679-532

Figure 176. Two-path polar operation.

bias winding is unnecessary, because the polar relays are operated from mark to space and from space to mark by the reversals of current flow. No send bias adjustment is required. The carrier telegraph terminal can be connected to either one or two repeaters. When two repeaters are used, the send loop is connected to one repeater and the receive loop to a second repeater. Figure 177 shows the carrier telegraph terminal connected to one repeater. Some components have been omitted from the repeater for the sake of simplicity. The teletypewriter at station A is the receiving teletypewriter and the teletypewriter at station B is the transmitting teletypewriter. With this arrangement, home copy is received because the receiving (selector) magnet is in series with the transmitting contacts at station B.

(2) Transmission of polar signals from the carrier terminal to the repeater is over the RL leg and the transmission of polar signals from the repeater to the carrier terminal is over the SL leg. The amount of current flow usually is adjusted to 30 ma at the end of the loop where the battery is supplied. If the current at the receiving point is too low for proper operation, the current flow at the transmitting end can be raised to 40 or 50 ma. The current flow in line A is varied by adjusting the RL RES potentiometer. To increase the current flow in line B the ADJ CUR potentiometer is adjusted.

c. Transmission of Signals From Station B Teletypewriter to Carrier Telegraph Terminal.

(1) *Transmitting a mark signal.* At the repeater, neutral signals from the teletypewriter at the local station B (fig. 177) operate the S relay to its mark and space contacts, applying equal voltages, −E and +E, of opposite polarity to the line. When a mark signal is transmitted, the S relay is operated to the mark contact, and 30 ma of current flows from the negative battery, −E, through the ADJ CUR potentiometer, line B, R78, the SEND and BK relays to ground, and returns through ground. The current is in a direction to operate the SEND and BK relays to the mark contact. The send relay allows the carrier tone to be transmitted to the distant carrier telegraph terminal.

(2) *Transmitting a space signal.* During a spacing interval, the S relay in the repeater is operated to the space contact and applies a positive potential, +E, to line B. As a result, 30 ma of current flows from the negative side of the +E battery through ground, the send and break relays, R78, line B, and the potentiometer, to positive battery +E. The current flow is now in a direction to operate the SEND and BK relays to the space contact. The SEND relay now causes the carrier terminal to transmit a space signal to the distant carrier telegraph terminal.

d. Transmission of Signals From Carrier Telegraph Terminal to Station Teletypewriter A.

(1) *Transmitting; a mark signal.* When the REC relay is operated to the mark contact, a current of 30 ma flows from the negative battery at the mark contact of the REC relay through L2, R70, RL RES, R77, the line A, and the R relay, to ground, and returns through ground. The direction of the current through the R relay is such as to cause it to operate to the mark contact. When the R relay is operated to the mark contact, mark current flows through the local station A. It flows from the −115-volt battery at station A through the transmitting contacts, receiving magnet, line potentiometer, the local wire, the mark contact of the R relay to ground, and returns through ground.

(2) *Transmitting a space signal.* When the REC relay at the carrier terminal is operated to the space contact, positive battery is applied to the loop. A current of 30 ma flows from the negative terminal of the battery at the REC relay through ground, the R relay, line A, R77, RL RES, R70, and L2 to the space contact of the REC relay to positive battery. The direction of current flow through the R relay operates it to the space contact. When the R relay is on the space contact, the local station circuit is opened and no current flows in the local circuit. In this manner the polar signals have

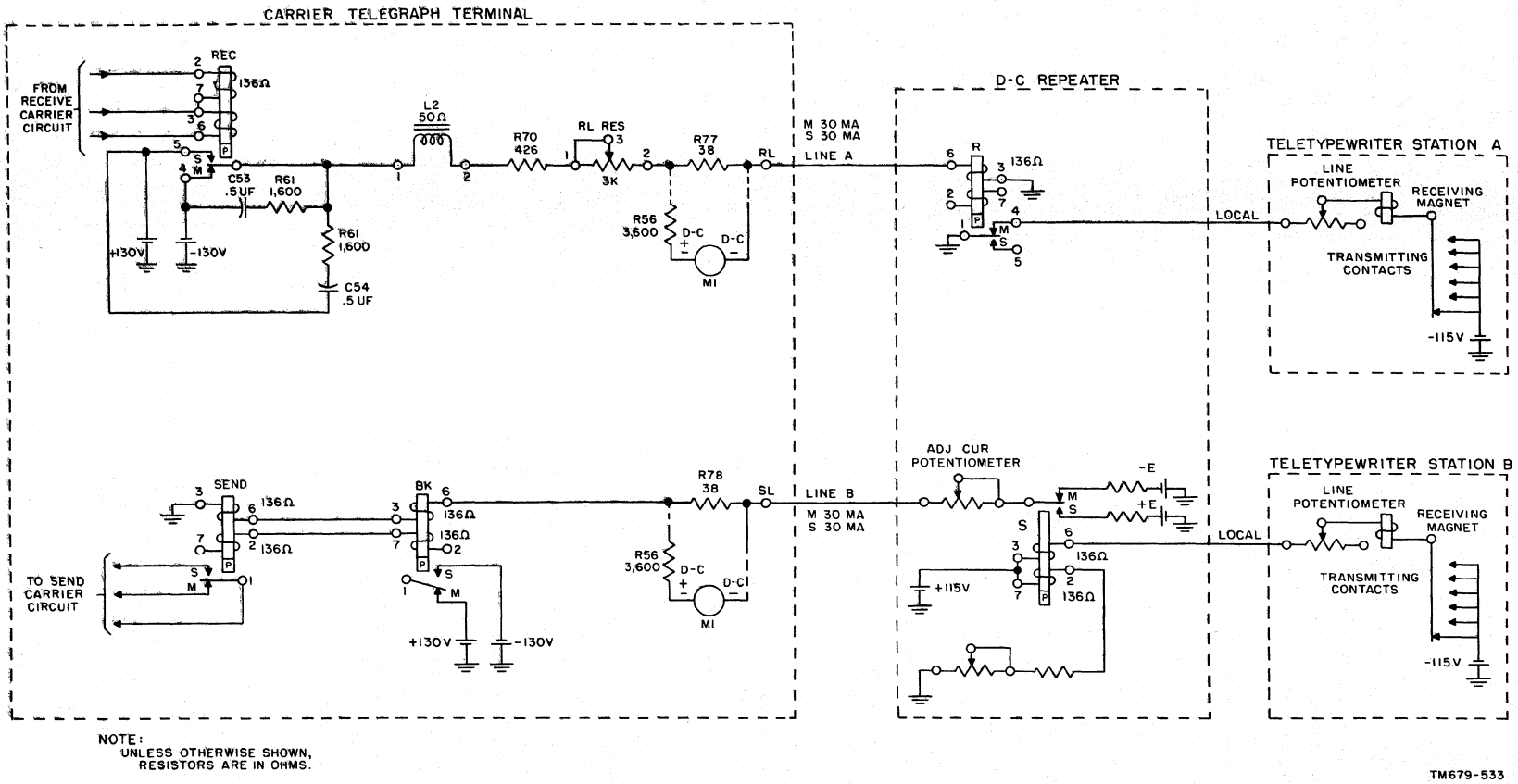


Figure 177. Functional diagram of two-path polar operation.

been converted to neutral ones, since current flows in the local circuit during the marking interval and no current flows during the spacing interval.

145. Polarential Operation (Half-Duplex)

a. General. A circuit arranged for *polarential* operation transmits *polar* signals in one direction and *differential* signals in the other direction. The combined effect is the equivalent of polar transmission in both directions on one telegraph circuit. These two words, polar and differential, have been combined to form the word *polarential*. Figure 178 illustrates a d-c repeatered circuit using polarential operation. Neutral signals are transmitted by the A teletypewriter to the A repeater which converts the neutral signals to polar signals and transmits them to repeater B. Here, the polar signals are changed to neutral signals and transmitted to the B teletypewriter. In the B to A direction, neutral signals are transmitted by the B teletypewriter to the B repeater which converts the neutral signals to differentials and transmits them to the A repeater. The A repeater converts the differential signals to neutral signals for reception by the A teletypewriter.

b. Circuit for Polarential Operation. A, figure 179, is a simplified circuit illustrating a repeatered circuit using polarential operation.

(1) *Polar sending.* At A, the polar sending end, two batteries of equal but of opposite polarity are applied to the mark and space contacts of the sending relay. The B, or differential sending end of the circuit, is terminated in ground when the armature of the sending relay at B is operated to mark. The operation is half-duplex because the differential sending end must be in an idling condition when A transmits; that is, the armature of the SEND relay at the differential end must be on the mark contact. The polar sending end transmits signals which are reversals of current flow. The REC relay at B operates on these reversals of current flow and requires no bias circuit. The REC relay at B closes the local circuit on mark and opens the local circuit on space. In the example shown, the magnitude of current is 18 ma in the line and thus in the line winding of the REC relay at A. The current in the bias winding of the relay is 32 ma in a marking direction. This prevents the REC relay at A from operating to space when a mark signal is transmitted at the polar sending end, and holds the relay more firmly to the mark contact when a space signal is being transmitted.

(2) *Differential sending.* In a differential transmission system, the marking and spacing currents are in the *same direction* but of *different magnitude*. When the sending relay at B is operated to mark (by a teletypewriter or connected loop being in a marking condition), ground is applied to the line over the mark contact. Current of 18 ma flows from the —75-volt battery at the mark contact of the send relay at A over the loop to ground at B. This current tends to operate the REC relay at A to the space contact, but the 32 ma of current in the bias winding holds the relay to mark. During a spacing interval, the SEND relay at B is operated to space, applying +115 volts to the line. The current flowing increases to 46 ma but is flowing in in the same direction as it was when the circuit was marking. The magnitude of current is now great enough to overcome the effect of the local biasing current and to operate the REC relay at A to space.

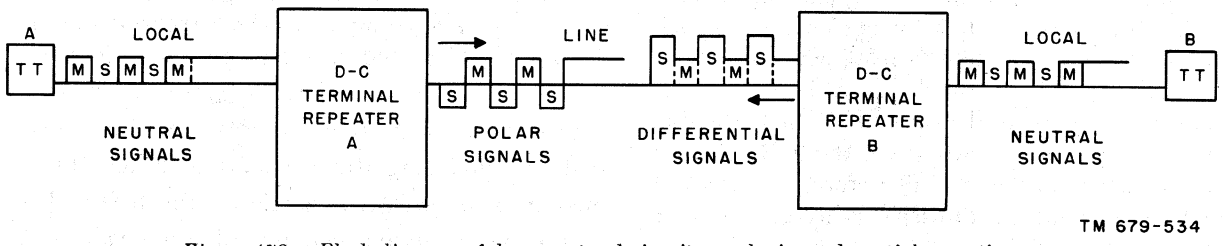


Figure 178. Block diagram of d-c repeatered circuits employing polarential operation.

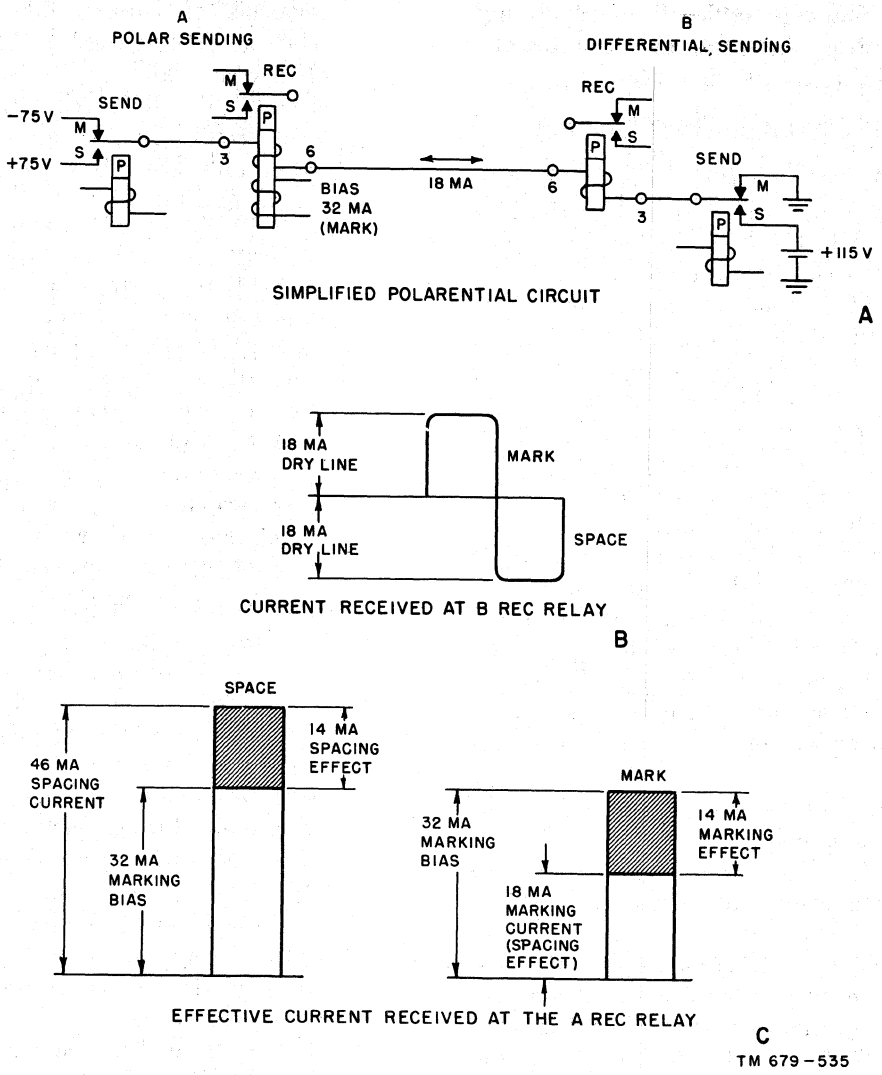


Figure 179. Theoretical polarential operation.

c. Waveshapes. B and C (fig. 179) show the polar and differential waveshapes. Diagram C shows that when a spacing signal is sent, the differential spacing current (46 ma) is greater than the marking bias (32 ma). The A REC relay is therefore operated to the space contact. When a marking signal is sent, the 32 ma marking bias is greater than the 18 ma marking current (which is in a direction to give a spacing effect on the A REC relay). As a result, the A REC relay is operated to the mark contact. The effective marking and spacing currents at the A REC relay is then 14 ma, as shown. Therefore, the effect of differential signals on the REC relay at the polar sending end is polar.

d. Effect of Line Leakage on Polarential Operation. The advantage gained by the use of polarential operation is that adjustments are not required to compensate for variations in electrical characteristics of the line caused by changing weather conditions. Leakage paths along the loop, which occur when the wire is wet, increase the marking current and decrease the spacing current received at the polar sending end when transmission is from the differential sending end. The increase in marking current and the decrease in spacing current under leakage conditions are of approximately equal magnitude, resulting in the transmission of undistorted signals by the REC relay at the polar sending end for all usual values of leakage.

146. Carrier Telegraph Terminal Arranged for Polarential Operation (Half-Duplex)

a. General. In figure 180, B is a block diagram of a system of carrier telegraph with one channel employing polarential (half-duplex) operation. The signals from the A teletypewriter are neutral toward the A repeater which transmits them as differential signals to the A carrier terminal. The A carrier terminal converts the d-c signals to a-c signals and transmits them to the B carrier terminal which converts the a-c signals to polar signals and transmits them to the B repeater. The signals are changed to neutral signals and are transmitted to the B teletypewriter. The transmission from the B to the A direction follows the same sequence of operations as from A to B. Transmission is in one direction at a time. One channel can employ polarential operation, while the other channels are using neutral operation. It is also possible to operate the A loop circuit of a channel on a polarential basis while the B loop circuit is operating on a neutral basis. However, if the A loop is operating on a half-duplex basis, the B loop also should operate on a half-duplex basis.

b. Transmission of Signals from Carrier Telegraph Terminal to Repeater.

(1) Figure 181 is a functional schematic of a loop operating polarential half-duplex. Transmission from the carrier telegraph terminal REC relay is on a polar basis. The line current flowing is 30 ma (adjusted by the SL RES). With the REC relay on the mark contact, the 30-ma loop current flows through the windings (3–6) of the SEND and BK relays through the loop to ground at the mark contact of the SEND relay in the repeater. The current is in a direction to cause the REC relay at the repeater to operate to mark. The current in the windings (3–6) of the SEND and BK relays is in a direction to cause them to operate to space. However, the biasing current, 47.5 ma through the 2–7 windings, holds them to mark.

(2) When the REC relay at the carrier terminal operates to space, the 30 ma of line current flow is reversed. The current flow in the 3–6 winding of the REC relay at the differential sending end (the repeater) is in a direction to cause it to operate to space. Thus, the signals are relayed to a local circuit or teletypewriter. The current in the 3–6 windings of the SEND and BK relays at the carrier terminal is in a direction to hold their armatures to mark. These relays must be on mark while the REC relay is transmitting to prevent false operation at the distant carrier telegraph terminal.

c. Transmission of Signals from the Repeater to the Carrier Telegraph Terminal.

(1) When transmission is from the station teletypewriter, the SEND relay in the repeater is operated by a teletypewriter or local circuit. The loop is terminated in ground during marking intervals and in +115 volts during spacing intervals.

(2) On mark, the current flows from the −130-volt at the mark contact of the REC relay at the carrier terminal through the loop to ground. The effect is the same as when a mark is transmitted from the polar transmitting end. The SEND and BK relays are held on mark by the current in the bias circuit, thus allowing carrier tone for a mark signal to go to the distant carrier telegraph terminal.

(3) During the spacing interval, the SEND relay at the repeater operates to space,

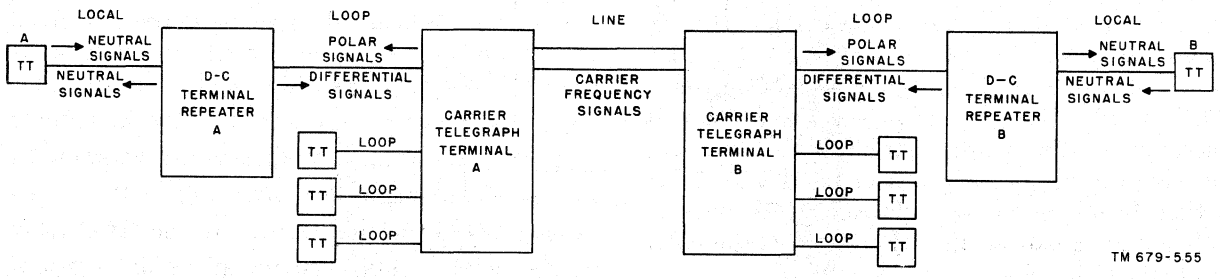


Figure 180. Block diagram of carrier telegraph terminal employing polarential operation.

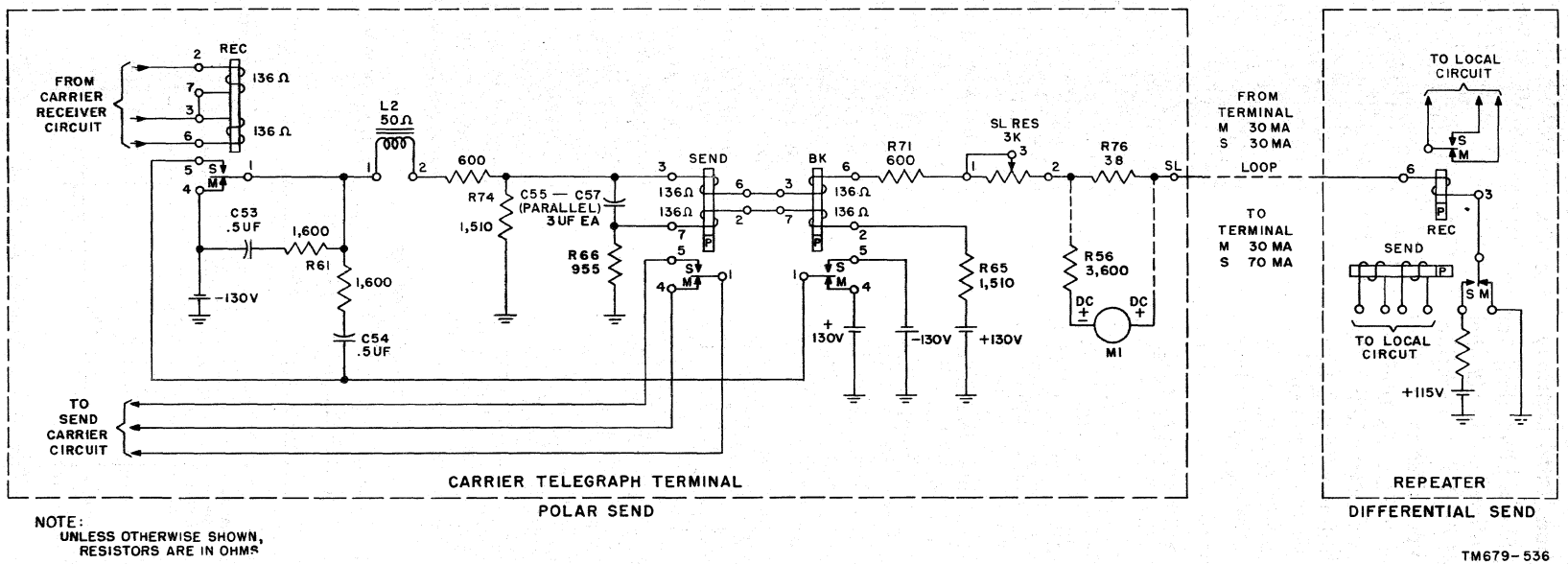


Figure 181. Polarential operation, half-duplex.

applying +115 volts to the loop. The current in the loop increases to approximately 70 ma and is in the same direction as it was when a mark was transmitted. The 70 ma is in a direction to cause the SEND and BK relays in the carrier terminal to operate to space. Since the current in the bias circuit is only 47.5 ma, the 70 ma operates the relays to the space contact. The SEND relay then permits a space signal to be transmitted to the distant carrier terminal.

147. Tandem Connection of Two Carrier Telegraph Terminals

a. General. A carrier telegraph channel loop terminal (SL) associated with a bay of one system can be connected on the d-c loop side to a channel loop terminal of another system. This connection provides for service between two stations, each associated with a different system. This arrangement is called a *tandem* connection, and the operation over this connection is called *through* operation. Figure 182 illustrates two systems with channels 3 and 4 connected in tandem. The channel loops are indicated as CHAN-3 and CHAN-4. Channels 1 and 2 may be half- or full-duplex operation as desired. Channel 3 is arranged for half-duplex operation; channel 4 employs full-duplex operation. Usually, the two terminals connected in tandem are located at the same place, which is all the *junction* point.

b. Half-Duplex Operation. Figure 183 is a schematic of the CHAN 3 loop circuits of carrier telegraph terminals 2 and 3 shown in figure 182 and illustrates two channels using neutral half-duplex operation, with the loops (of terminals 2 and 3) connected in tandem. The send leg of a channel of one system is connected to the send leg of a channel of the second system. One loop operates to negative battery while the second loop operates to positive battery. This is necessary, because series aiding battery is required at either end of the loop. For operation of a neutral half-duplex loop, see paragraph 140.

(1) *Transmission of signals from carrier telegraph terminal 2 to carrier telegraph terminal 3.* The operation is half-duplex; therefore, the REC relay at terminal 3 is held on mark by a mark signal from the REC carrier circuit. When the REC relay at terminal 2 is on the mark contact, 60 ma of current flows in the loop from the negative battery at the mark contact of the REC relay at terminal 2 to the positive battery at the mark contact of the REC relay at terminal 3. The current is in a direction to hold the SEND and BK relays at terminal 3 to the mark contact. The SEND relay on the mark contact allows the carrier mark signal to be transmitted to the line to terminal 4. When the REC relay at terminal 2 is operated to space, the battery applied at either end of the loop is of the same polarity. Very little or no current flows in the loop wire. The SEND and BK relays at terminal 3 are operated to space by the current flowing in the 2–7 bias windings. When the send relay is on space, a space signal is transmitted to terminal 4. The SEND and BK relays at terminal 2 are held to the mark contact while the REC relay at terminal 2 is operated to the space contact because the current flow through the bias windings of these relays is in a marking direction.

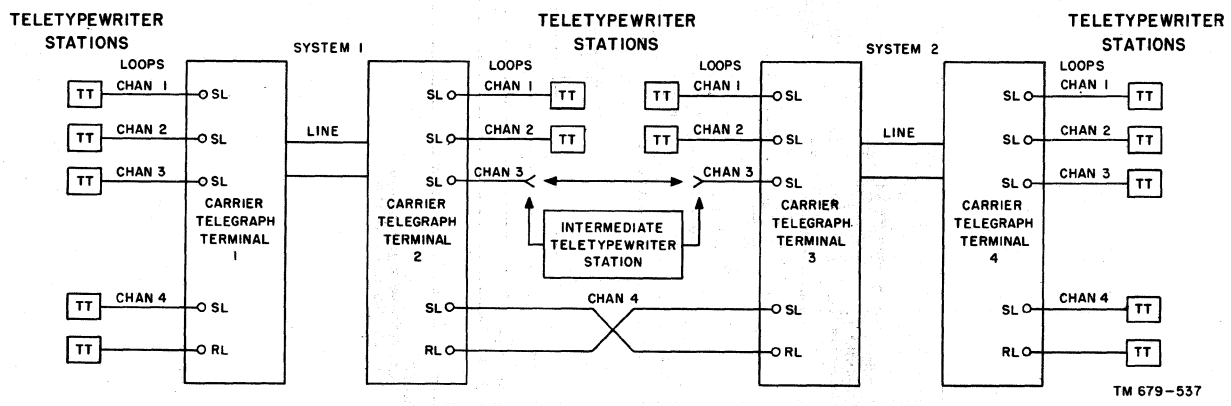


Figure 182. Tandem connections.

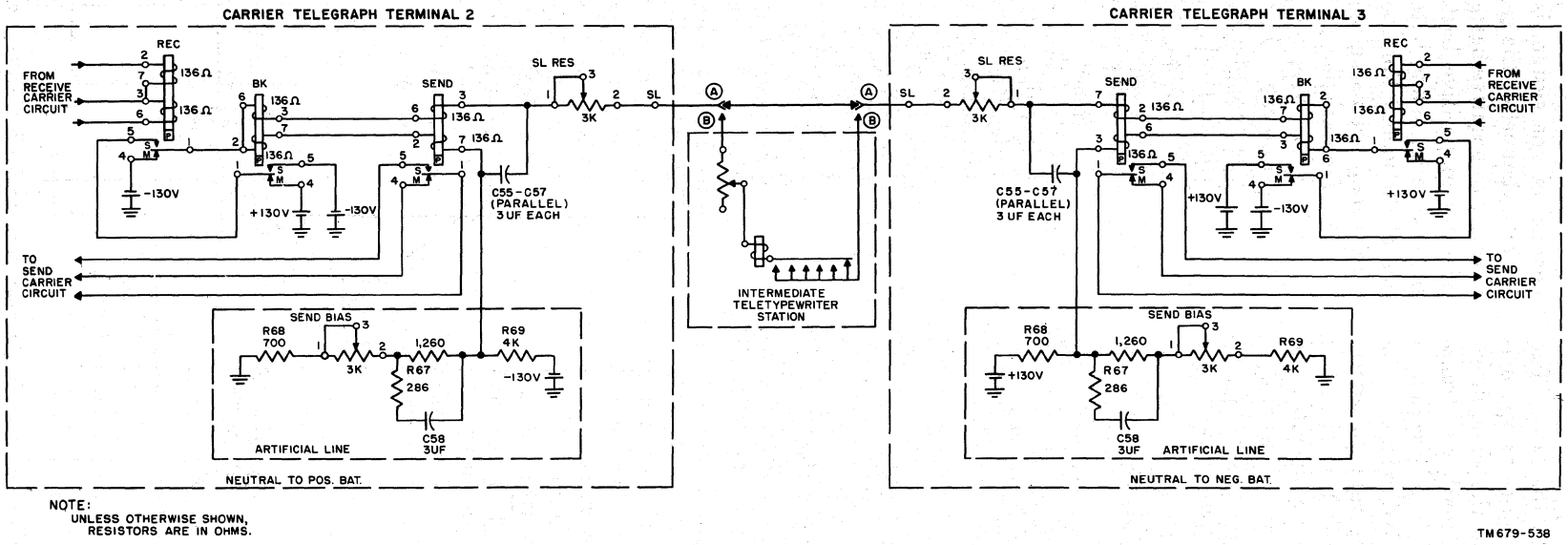


Figure 183. Functional diagram of through-connection, neutral, half-duplex operation.

(2) *Transmission of signals from carrier telegraph terminal 3 to carrier telegraph terminal 2.* The REC relay at terminal 2 must be on the mark contact since this is half-duplex operation. When the REC relay at terminal 3 is operated to mark, 60 ma of current flows in the loop circuit in a direction to operate the SEND and BK relays at terminal 2 to mark. The SEND relay at terminal 2 allows a mark signal to be transmitted over the line to terminal 1. When the REC relay at terminal 3 is operated to the space contact, little or no current flows in the loop wire because the batteries across the loop are series opposing. The current in the bias windings of the send and break relays at terminal 2 operates them to the space contact. The SEND relay causes a space signal to be transmitted to terminal 1.

(3) *Intermediate teletypewriter.* The connection between the two carrier telegraph terminals may be direct, as shown by the A wiring option; or a teletypewriter may be inserted into the loop as shown by the B wiring option. This teletypewriter can receive from either terminal 2 or terminal 3, but transmits simultaneously to both terminals. Since the signals in the loop are neutral, the receiver magnet of the teletypewriter responds to them. When the teletypewriter is transmitting a mark, a transmitter contact closes the loop, allowing current to flow. The loop current operates both SEND and BK relays to mark, allowing a mark signal to be transmitted in both directions. When the transmitter contacts are open, the loop is open and no current flows in the loop. The send and break relays at both terminals are oprated to space by the current in the bias windings; therefore, a space is transmitted in both directions.

(4) *Bias adjustment.* When the send bias adjustment is made at carrier telegraph terminal 2, repeated spaces must be sent from terminal 3 to terminal 2. If the send bias adjustment is being made at terminal 3, then repeated spaces must be sent from terminal 2 to terminal 3. However, the bias adjustment may be made at either terminal when the teletypewriter at the junction sends repeated spaces. Alternate R and Y signals sent

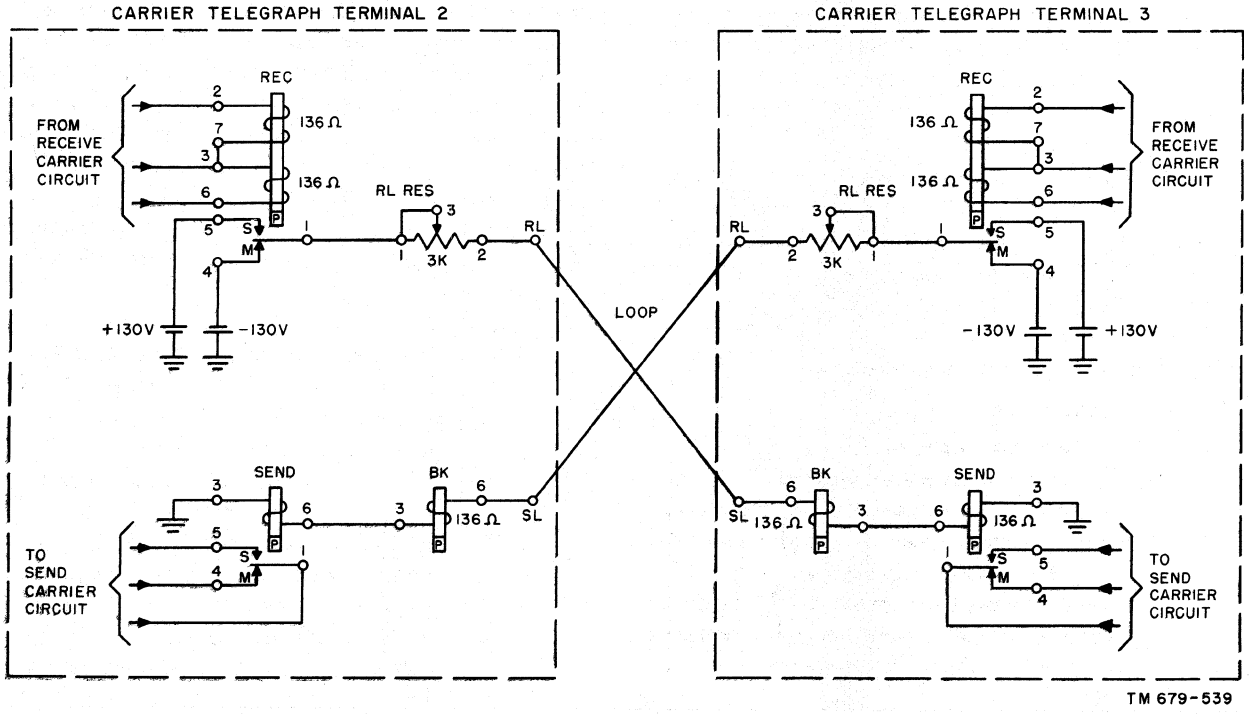


Figure 184. Functional diagram of through-connection, two-path polar operation.

from a tape or dots from a test set may be required instead of repeated spaces.

c. Full-Duplex Operation. CHAN-4, figure 182, illustrates a tandem connection operating on a full-duplex basis. The send leg of the channel of one terminal of one system is connected to the receive leg of a channel of the terminal of the second system and vice versa. This operation may be neutral or polar full-duplex. Figure 184 illustrates two channels connected in tandem employing two-path polar (full-duplex) operation. Two-path polar operation is described in paragraph 144. Since the operation is polar, a bias adjustment is unnecessary.

(1) *Transmission from carrier telegraph terminal 2 to carrier telegraph terminal 3.* When the REC relay at terminal 2 is operated to the mark contact, negative battery is applied to the loop. Current flows from the negative battery over the RL leg of terminal 2 through the SL leg of terminal 3 to ground. The SEND and BK relays at terminal 3 are operated to the mark contact. The SEND relay allows a mark signal to be transmitted to terminal 4. When the REC relay at terminal 2 is operated to the space contact, positive battery is applied to the loop, causing the current flow in the loop to reverse. The SEND and BK relays in terminal 3 are operated to the space contacts and a space signal is transmitted to terminal 4.

(2) *Transmission from carrier telegraph terminal 3 to carrier telegraph terminal 2.* The transmission of signals from the REC relay at terminal 3 to terminal 2 is identical with that described above.

148. Summary of Loop Circuits

a. Loop Operations.

(1) A telegraph communication network is set up by connecting several teletypewriter stations by means of switchboards, repeaters, and carrier telegraph terminals, with intervening transmission lines. The circuit connecting two switchboards is called a trunk. Loop circuits connect any two adjacent elements in the network.

(2) The loops may be d-c or v-f (voice-frequency) loops. With most of the Army systems in the field only d-c loops may be used. The newer equipment can be arranged for either d-c or v-f loop operation.

(3) Loop operations may be grouped as neutral, polar, or polarential; half-duplex or full-duplex. Neutral operation may be to positive or negative battery.

b. Neutral Operation.

(1) In neutral operation, the power supplies of the elements at opposite ends of the line, as at station and terminal, are connected in series aiding for marking pulses, and in series opposing for spacing pulses. Operation is said to be *to positive battery* when the positive terminal of the station power supply is connected to the line.

(2) Neutral operation may be half-duplex or full-duplex. In half-duplex operation, a single transmission line permits transmission in either direction, but not in both simultaneously. In full-duplex operation, two lines are used, permitting simultaneous transmission in both directions.

c. Two-Path Polar Operation. A two-path polar loop provides full-duplex operation. It employs two transmission lines. Marking and spacing pulses are of equal amplitude but opposite polarity. The power supply is applied at one end of the loop.

d. Polarential Operation. A polarential loop provides half-duplex operation over a single transmission line. For transmission in one direction, polar signals are sent. For transmission in the opposite direction, differential signals are used. In differential transmission, marking and spacing pulses are of the same polarity, but of different amplitudes.

e. Tandem Connections.

(1) When the loop side of a carrier telegraph terminal is connected to the loop side of a second carrier telegraph terminal, the terminals are connected in tandem. The circuit is called a through circuit because it goes through two carrier systems.

(2) The operation may be half-duplex or full-duplex. For half-duplex operation, the SL binding post of one terminal is connected to the SL binding post of the second terminal. For full-duplex operation, the SL binding post of the first carrier terminal is connected to the RL binding post of the second carrier termi-

nal, and the RL binding post of the first carrier terminal is connected to the SL binding post of the second carrier terminal.

(3) The operation may be neutral or polar. For neutral operation, the loop of one carrier terminal operates *to negative battery* while the loop of the second carrier terminal operates *to positive battery*. This arrangement is necessary to have the polarity of the batteries in series aiding for a mark signal. In polar operation, the two carrier terminals operate on a two-path polar basis.

149. Review Questions

a. List six types of d-c loop operation used by Army carrier telegraph terminals.

b. What precaution must be observed when connecting a polar relay into a circuit?

c. What is the polarity of the battery applied to the loop at the teletypewriter station when the loop is neutral operation to negative battery? What polarity of battery is applied to the mark contact of the REC relay for this loop?

d. Why must a *send bias adjustment* be made when neutral operation is employed?

e. Which polar relay controls the oscillator output?

f. Is it possible to obtain home copy when the loop operation is on a full-duplex basis? Explain your answer.

g. What is the purpose of the BK relay when neutral half-duplex operation is used? When neutral full-duplex operation is used?

h. Why must the REC relay be operated to the mark contact when the local typewriter is transmitting when the operation is half-duplex?

i. What is meant by *"breaking"*?

j. When is a *full-metallic* loop used?

k. Is a send bias adjustment necessary when two-path polar (full-duplex) operation is used? Why?

l. Why is polarential half-duplex operation used rather than neutral operation when there are leakage paths along the loop?

m. What types of signals are transmitted over a polarential loop?

n. Why is the current in the bias windings of the SEND and BK relays in a direction to operate them to mark when polarential operation is used?

o. What is the purpose of connecting two carrier telegraph terminals in tandem?

Section IV. ELEMENTS OF CARRIER-TELEGRAPH SYSTEMS

150. Fundamentals of Carrier Telegraphy

a. Purpose of Carrier Telegraphy. In d-c telegraphy the major advantage of full-duplex operation over half-duplex operation is that full use is made of the message-carrying potentialities of the circuit. Over a half-duplex circuit, only a single message can be sent in one direction at any one time; a full-duplex circuit can handle messages in both directions at the same time. In a carrier-telegraph system, it is possible to transmit many messages simultaneously. Furthermore, the use of a carrier-telegraph system extends the distance over which messages can be transmitted, in addition to increasing the message-carrying capacity of a transmission line, and therein lies its importance for the Army communication system. The basic circuit of a carrier telegraph system is illustrated in figure 185. For the sake of simplicity, filters at the terminals have been omitted. The equipment is so arranged that only those channels having the same number can operate together. Thus, channel 1 at terminal A operates with channel 1 at terminal B, channel 2 with 2, and so on. Messages are transmitted simultaneously from terminal A to terminal B over the four channels, each at a different frequency. The frequencies used for the different channels must be separated widely enough to prevent their interfering with one another during transmission. The messages sent from terminal B to terminal A are also transmitted at different frequencies. In addition, the four frequencies used for transmission from A to B are different from the four frequencies used for transmission from B to A. This is done to eliminate the possibility of interference when both stations are sending simultaneously.

b. General Description of Carrier System. The elementary schematic of a carrier telegraph set, shown in figure 186, illustrates the essential components of such a set. Fundamentally, a carrier telegraph system uses a relay circuit operating from d-c pulses just as in d-c telegraphy. However, this relay alternately opens and closes the circuit to an oscillator which is the source of an

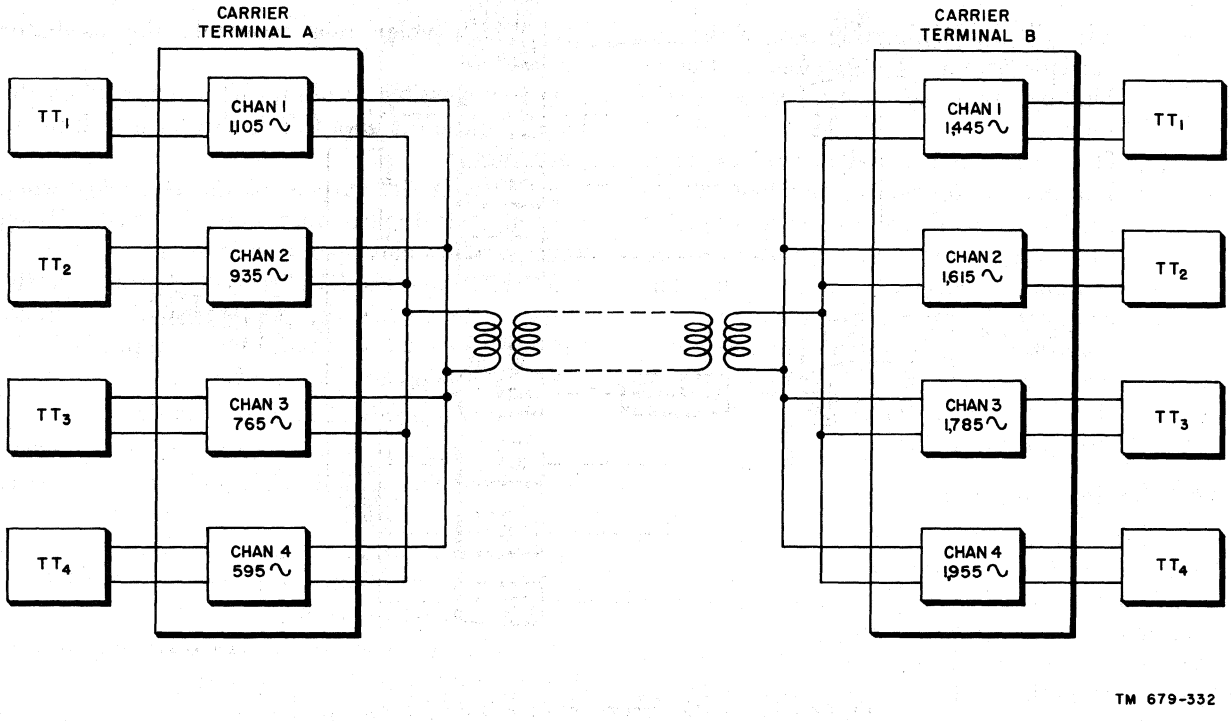


Figure 185. Principle of multi-channel carrier-telegraph system.

a-c signal of a definite frequency. There is a separate oscillator for each transmitting channel. The impulses of alternating current from the oscillators are then impressed on the single transmission line over which the signal is to be transmitted. Before being impressed on the line, however, the signal is first passed through a channel filter, to remove any frequency components that might interfere with adjacent channels. At the input to the receiver, there must be a means of separating the various messages for the different channels (fig. 186). The selecting filter for receiver-channel 1 permits only the signal from transmitter-channel 1 to pass through to the detector for channel 1. The selecting filter for channel 2 provides the signal for the detector of channel 2, and so forth. At the detectors, the signal is demodulated; that is, the a-c impulses are reconverted into the original d-c impulses. This reconversion is necessary to operate the teletypewriter receiving mechanism at the receiving station. From this brief description of the operation of a carrier telegraph set, it can be seen that the steps required for accomplishing the total result may be summarized as follows:

(1) Vacuum-tube oscillators, or oscillators of some other type, provide the currents of different selected frequencies to be used as *carriers*.

(2) The message impulses from the teletypewriter are impressed upon the carrier. This process is called *modulation*.

(3) The separate modulated carrier currents at the receiving end are separated by means of selecting circuits known as filters.

(4) The message impulses are separated from the carrier for transmission to, and operation of, the receiving teletypewriter. This process is called *demodulation*.

151. Types of Carrier Telegraph Systems

a. Single-Channel Systems.

(1) The different carrier telegraph systems in use for the transmission of signals over a single channel of a teletypewriter circuit differ from one another, basically, in the manner in which the carrier is modulated —that is, the manner in which the message signal is superimposed on the carrier. Five methods of modulation, all using carrier frequencies in the voice-frequency range, are on-off; single-side-

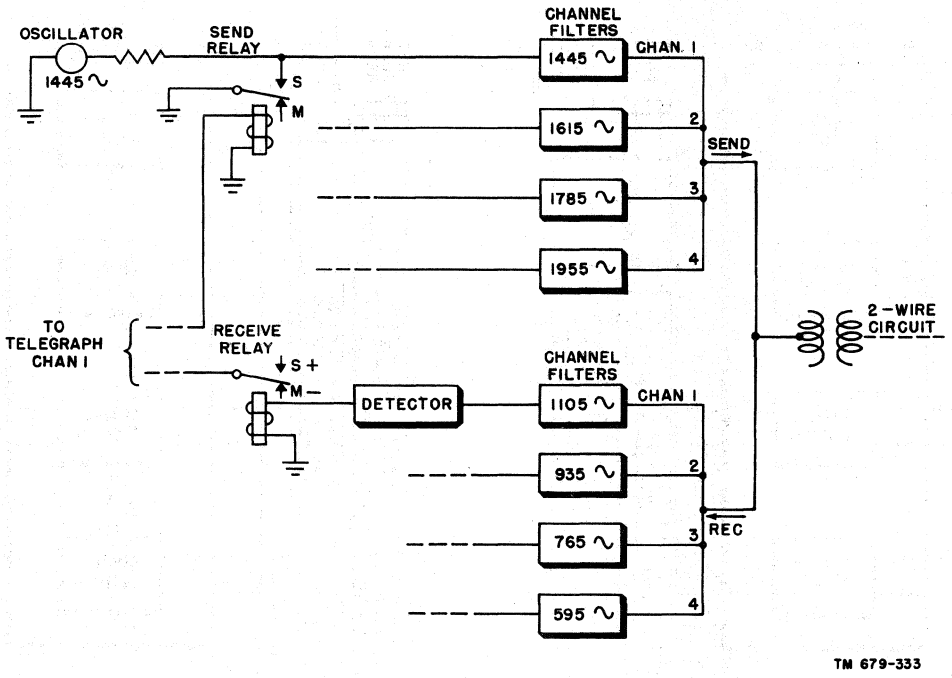


Figure 186. Partial schematic of carrier-telegraph set.

band; two-source; frequency-shift; and one-source two-band. The first two methods use amplitude modulation; the last two use frequency modulation; and the third, the two-source method, uses amplitude modulation, but the resulting signal is similar to a frequency-modulated signal. The difference in modulation methods determines the type of detector used to demodulate the signal at the receiving station. For the first two types, the detector must be capable of detecting amplitude modulation. The last two types require a discriminator, capable of detecting frequency-modulated signals. The two-source method can use either kind of detector. The two most widely used types of modulation are the on-off method and the frequency-shift method of operation. Because of technical advantages of the frequency-shift system, it has been adopted for the newly designed systems.

(2) As another point of difference, frequency-modulated systems are far less susceptible to interference from noise currents than amplitude-modulated systems. As a result, the signals transmitted in frequency-modulated systems are less liable to distortion. For this reason, Army use of frequency-modulated carrier-telegraph systems is increasing, while use of the earlier amplitude-modulated systems is declining.

b. On-Off Method.

(1) Block diagrams and bandwidth characteristics of five voice-frequency telegraph systems are shown in figure 187. In the *on-off method*, in A, the presence of carrier current on the line corresponds to a marking pulse. The absence of any carrier current on the line corresponds to a spacing interval. The signal is formed by the on-off relay modulator, which acts as a switch connecting the carrier oscillator to the line for marking pulses, and breaking the circuit for spacing intervals. The operation of the relay is controlled by signals from the teletypewriter keyboard in the sending loop. At the receiver, the marking pulses of carrier current are converted back into d-c impulses by the amplifier detector. The d-c pulses, after being amplified, operate the receiving polar relay.

(2) The on-off method utilizes a single carrier frequency (f_c in the filter-loss diagram shown in A), normally located at the

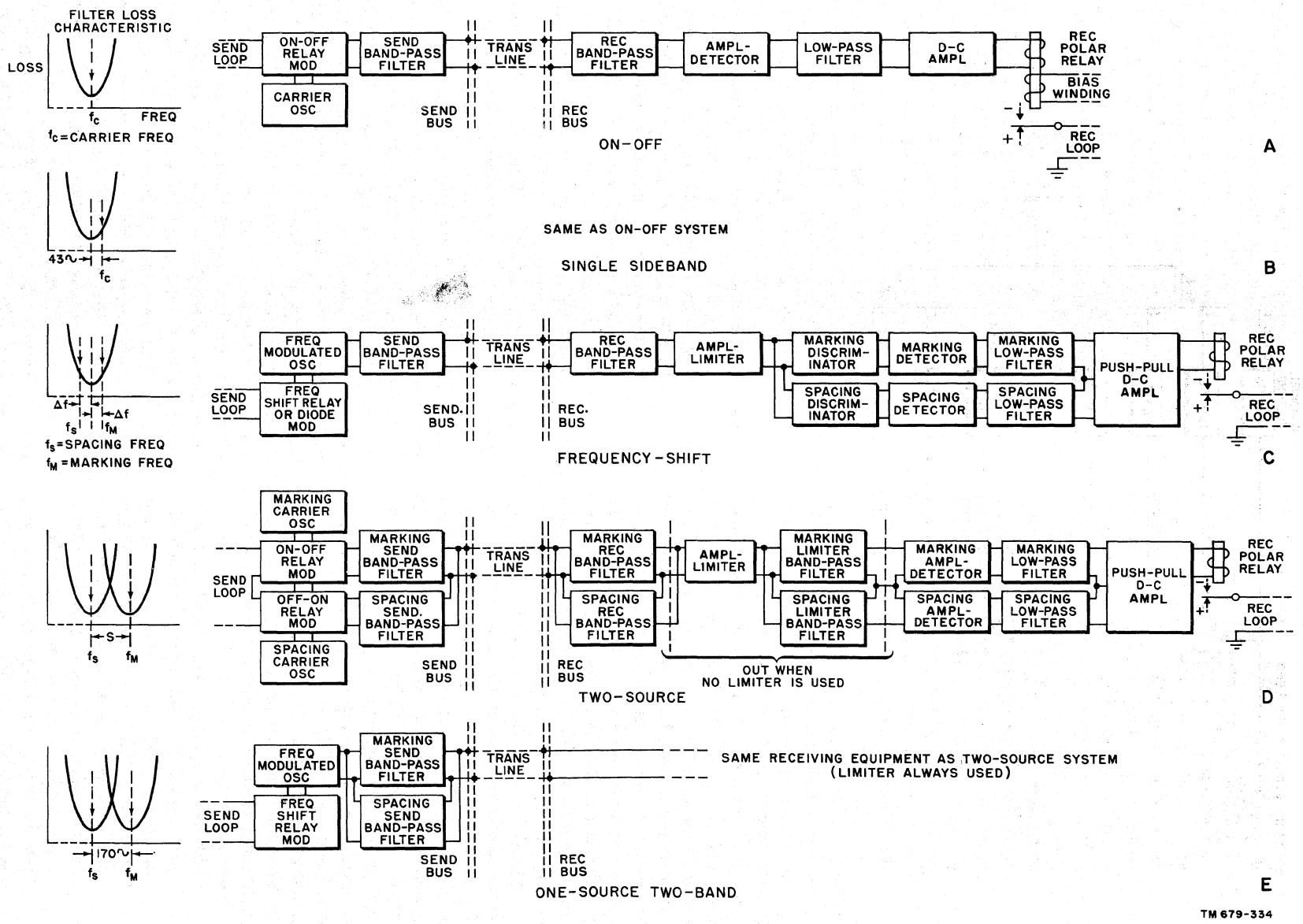


Figure 187. Voice-frequency telegraph systems.

center of the channel pass-band. An amplitude-modulated signal is the sum of a signal at the carrier frequency, and two signals (sidebands) with frequencies above and below the carrier frequency. In carrier telegraphy, the bandwidth required for the transmission of the carrier and both sidebands is about 85 cycles. However, the carrier frequencies used in a single set are spaced 170 cycles apart (fig. 186). The attenuation characteristic of the sending band-pass filter must be such as to cause minimum attenuation of the carrier and sidebands at the center of the channel, and maximum attenuation of the remainder of the 170-cycle channel. This insures minimum interference with adjacent channels.

c. Single Sideband Method. The circuits required at the sending and receiving ends of the line using the single sideband method of carrier-telegraphy operation (B, fig. 187) are exactly the same as those used for the on-off method. Also, marking pulses in both methods are transmitted as pulses of a-c current, and spacing intervals as intervals of no current. In the *single sideband method*, however, the carrier frequency is placed above the center of the band-pass channel of the filter. This off-center displacement of the carrier frequency causes the sideband with the frequency above the carrier to move upon the attenuation curve, whereas the sideband with the frequency below the carrier remains on the minimum-attenuation portion of the curve. The result is that the upper sideband is attenuated so much more than the lower sideband during transmission that it is effectively suppressed. This reduces the frequency bandwidth required for the transmission of the carrier signals, and decreases the possibility of interference with adjacent channels. During the spacing condition, carrier current may be either absent from the line or, if present, with an amplitude less than that of the marking current.

d. Frequency-Shift Method.

(1) The *frequency-shift method* utilizes a carrier signal of substantially constant amplitude from a frequency-modulated oscillator. The instantaneous frequency of the carrier signal varies between two limits, which correspond respectively to the marking and spacing conditions of the channel. On the loss diagram shown in C, figure 187, the two frequencies are designed as f_s for the spacing frequency and f_m for the marking frequency. The frequency limits are symmetrically located about the center of the transmission band. The transition between the limits may be abrupt or gradual. Attenuation by the sending filter must be minimum for the band containing the marking and spacing frequencies, and maximum outside this band.

(2) The block diagram of the circuit for the frequency-shift method shows that the carrier oscillator is controlled by a relay which, in turn, is controlled by signals from the teletypewriter keyboard in the sending loop. This relay does not act as a switch, however. The oscillator is always connected to the transmission line, and the relay changes the frequency of the carrier signal applied to the line. At the receiver, the detector must be capable of converting the frequency variations between spacing and marking frequencies into the d-c pulses necessary for operating the receiving teletypewriter. A discriminator is used as the detector in frequency-modulation circuits for converting frequency variations back into the original signal message. Separate marking and spacing discriminators are used to reduce distortion of the pulses.

e. Two-Source Method. Basically, the *two-source method* is a frequency-shift method, in that separate marking and spacing frequencies are used. However, the method of producing the two frequencies is quite different from that used in the frequency-shift method. Two on-off transmitters are used, each having a separate oscillator of a different frequency, as in D, figure 187. The sending relays of the two transmitters have their operate windings connected in series. The armature and contacts of the relays are connected to the oscillator circuits so that one oscillator delivers current to the line during marking intervals, the other during spacing intervals. The marking and spacing signals are confined to separate frequency bands. The receiving equipment usually consists of two on-off receivers with their output circuits differentially interconnected; that is, marking pulses cause current to flow through the REC relay in one direction and spacing pulses cause current to flow through the REC relay in the opposite direction. In this way, polar signals are

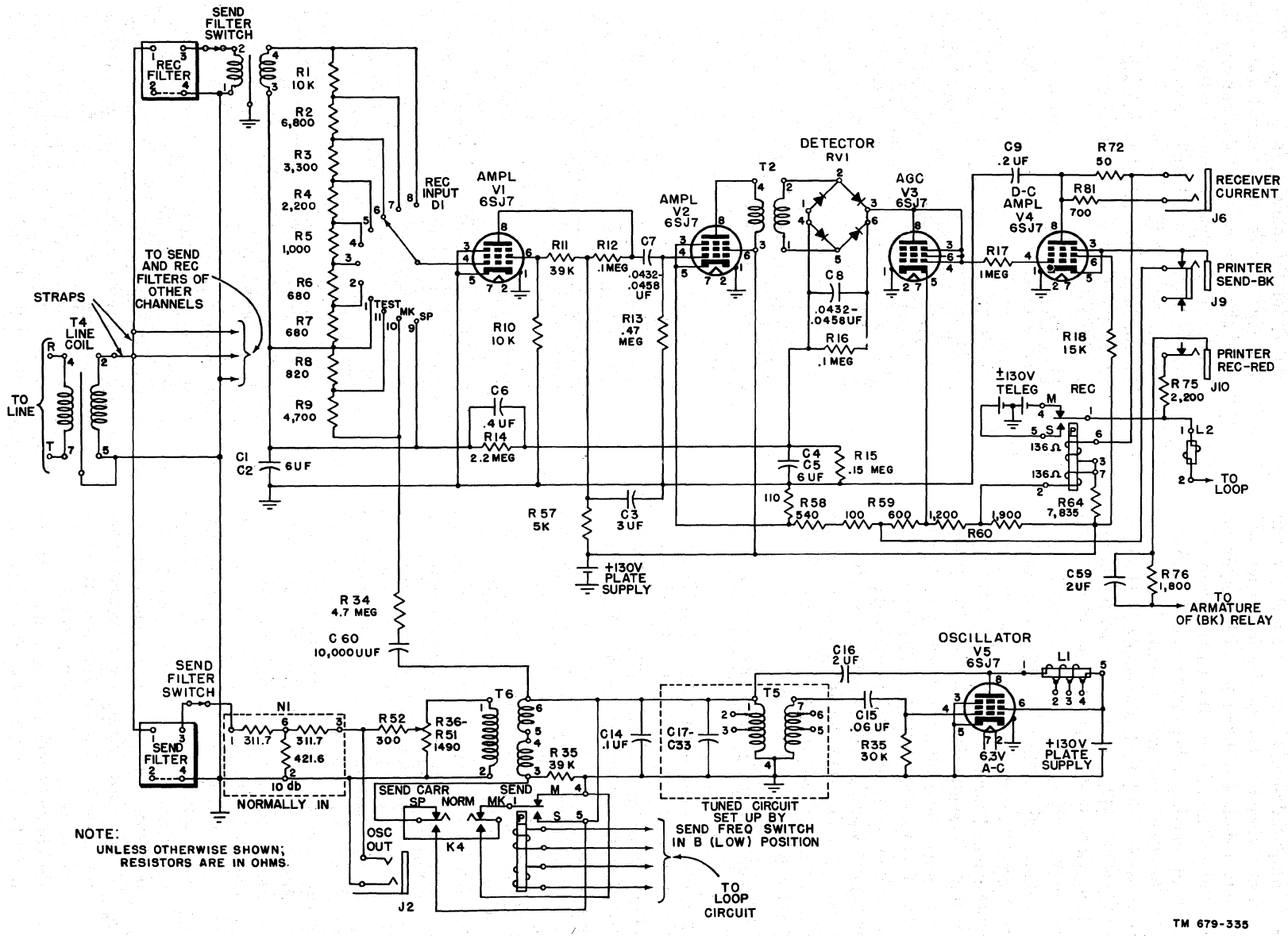


Figure 188. Carrier-telegraph terminal, sending and receiving circuits.

obtained for the operation of a common relay. The frequency transitions, in this method, are always discontinuous. As a result, the marking and spacing signals do not have phase continuity. This phase shift results in greater distortion than in a true frequency-shift system.

f. One-Source Two-Band Method. The *one-source two-band method* also is a frequency-shift method. The block diagram for the circuit used in this method, shown in E, shows that a single frequency-modulated oscillator is used at the sending end. Two separate filters are used. Through one filter, the marking frequency is passed to the transmission line. Through the other filter, the spacing frequency is passed to the transmission line. All other frequencies are rendered negligible because of attenuation. At the receiving end of the line, the circuit must be the same as that used with the two-source system. This method is used when a wide-band channel is desired and only narrow-band filters are available.

152. Elements of Amplitude-Modulated System

a. Figure 188, a simplified diagram of the sending and receiving circuits in a typical carrier-telegraph set, aids in understanding the details of a carrier-telegraph system using amplitude modulation.

b. This telegraph set uses eight carrier frequencies in the voice-frequency range (fig. 188). Two frequencies are assigned to each channel, one for transmission in each direction. The four frequencies used for transmission in each direction are in

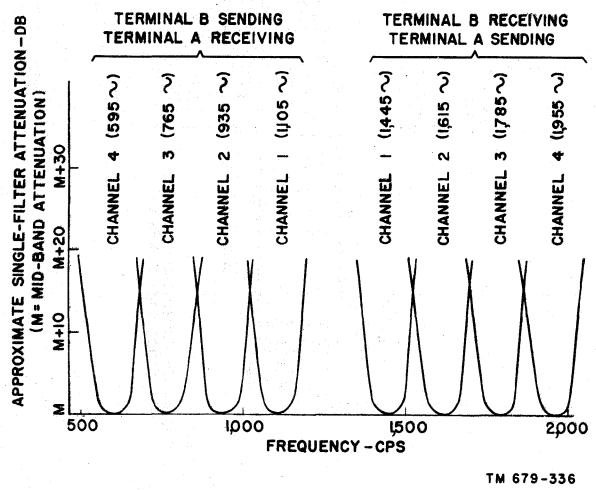


Figure 189. Allocation of carrier-telegraph frequencies and single filter characteristics.

a group, and are spaced 170 cycles apart. Figure 189 shows the allocation of these frequencies.

c. The elements of the telegraph set fall into two groups—the sending-circuit elements, consisting of the oscillator, the modulator, and the sending filter; and the receiving-circuit elements, consisting of the receiving filter, the receiving amplifier and automatic gain control system, the detector, and the d-c amplifier.

153. Sending Circuits

a. Oscillator.

(1) The *oscillator* of each channel is adjusted to send either of the two frequencies assigned to the channel (fig. 188). There are also 2 filters for each channel, 1 of which may serve as a sending filter while the other serves as a receiving filter. A switch, marked SEND FREQ (not shown), controls the selection of the oscillator frequency. A second switch, marked SEND FILTER, controls the selection of the filters as sending or receiving filters. If, at one terminal of the system, both switches are set on the position marked HIGH, each of the four oscillators sends out the higher of the two frequencies assigned to its respective channel; the filter of higher frequency becomes the sending filter and the filter of lower frequency becomes the receiving filter. The other terminal of the system must use the lower frequency for sending, and the higher frequency for receiving.

(2) The oscillator for one channel is shown in the lower right corner of figure 188. V5 is the oscillator tube. The resonant circuit consists of winding 1–4 of transformer T5 and tuning capacitors C17 to C33. These are the tuning capacitors for low frequencies. There is a different group of tuning capacitors for the high frequencies. A transient current in V5, resulting mainly from shot effect, is amplified and causes an a-c voltage to appear across the tuned circuit. The alternating current in the tuned circuit induces a voltage in winding 4–7 of transformer T5. This alternating voltage is applied to the grid of tube V5 through capacitor C15, is amplified, and through C16 is fed back to the tuned circuit in

the correct phase to sustain the oscillation. The oscillations increase in amplitude until the grid swings sufficiently positive to cause a flow of grid current (from cathode to grid). This grid current charges capacitor C15 during the positive part of the cycle. During the negative part of the cycle, C15 discharges through R33. The combination of C15 and R33 thus makes the grid negative with respect to the cathode. The negative bias limits the gain of the tube to such value that the oscillator output becomes constant. The output is applied across the primary of transformer T6. The output of the oscillator is called the *carrier*.

b. Modulator. In a carrier telegraph system, alternating current, supplied by an oscillator, is connected to the transmission line through a band-pass filter. In an on-off system, the sending relay, operated by d-c signals received from the sending loop, short-circuits this source of current during the spacing intervals, and permits the current to go to the line for marking signals. The telegraph signals then are said to *modulate* the voice-frequency carrier. Figure 190 shows the carrier pulses that are applied to the transmission line in an on-off system. The carrier current, flowing in winding 3–6 of transformer T6, is modulated by the operation of the SEND relay. When the SEND relay is on marking contact M, resistor R35 is short-circuited, so that the maximum carrier current flows in the circuit. When the armature is in midposition between the M and S contacts, resistor R35 is inserted in series with winding 3–6 of transformer T6. As a result, the carrier current flowing through the 3–6 winding decreases, and the power level of the carrier output (on the secondary side of T6) drops to approximately one-quarter the level of the marking output. When the SEND relay is operated to spacing contact S, winding 3–6 of transformer T6 is short-circuited and no current is transmitted to the line. Thus, when the SEND relay is operated, surges of carrier current, corresponding to the marking pulses in the loop circuit, are transmitted to the carrier line circuit. The potentiometer formed by tapped resistors R36 to R51 is adjusted to supply the desired carrier signal level at the output of the sending filter. The N1 pad provides a 10-db loss. This pad is removed from the circuit, by a switching

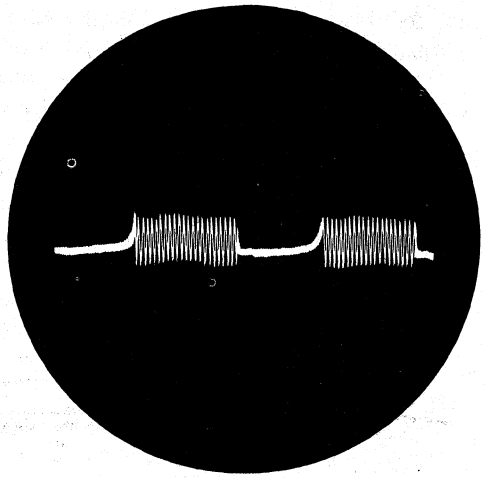


Figure 190. Carrier pulses applied to line in a-m system.

action not shown, when a high signal level is required. The jack (marked OSC OUT) is used to measure the output of the oscillator.

c. Filters. There are two *filters* in each channel of this telegraph set (fig. 188)—a sending filter and a receiving filter. A *sending filter* is used to suppress all sideband frequencies which are not necessary for the satisfactory transmission of signals in each channel. The function of the *receiving filter* is to separate the carrier and sidebands pertaining to each channel from all the others, and to guide them into the proper receiving circuits. The characteristics of the filters in the different channels are shown in figure 189. The attenuation characteristics are such as to pass the band of frequencies covering the carrier and those sidebands required for satisfactory transmission. The attenuation must increase as the frequencies lying outside the pass band approach the edges of the channel. In this way, the filter suppresses all the sideband frequencies that might interfere with the adjacent channels.

154. Receiving Circuits

a. Receiving Amplifier and Automatic Gain Control.

(1) The modulated carrier signals received from the line pass through line coil T4 to the receiving filters of all channels (fig. 188). The receiving filter of any channel causes a high attenuation of all frequencies except those near the center of the channel bandwidth. Each incoming modulated carrier signal is

directed by the filters into a separate receiving circuit or channel and applied across a tapped chain R1 to R7, which are used as an a-c voltage divider. These signals are amplified by a two-stage amplifier. The RECEIVE INPUT switch (D1) thus controls the magnitude of the signal voltage applied to the two-stage amplifier.

(2) The output of tube V1 is developed across plate load resistor R12 and coupled to the grid of tube V2 by the R–C (resistance-capacitance) combination of R13 and C7. Tube V2 amplifies the carrier signal to a suitable level for rectification. Output transformer T2 couples the output of tube V2 to the copper-oxide rectifier (varistor) RV1.

(3) Tube V3 provides an agc (automatic gain control) for the receiving circuit (fig. 188). The tube is connected as a diode, with the plate and grids tied together, and the combined plate is connected to the output of copper-oxide rectifier RV1. The cathode of V3 is biased sufficiently positive by the voltage drop across R58 and R59 to make V3 incapable of conducting until the plate voltage—which is the output of the rectifier—goes more positive than the cathode bias. If the level of the rectifier output is such that the plate voltage does exceed the cathode voltage, the resulting plate current develops a negative d-c voltage across the R–C combination of R15 and C4 and C5. This voltage is fed back to the control grid of tube V1, causing the grid to become more negative, and reducing the gain of this amplifier stage of the circuit. In this manner, the average plate voltage of tube V3 is kept fairly constant at a value near zero with respect to the cathode, unless the input level to the receiver is not high enough, even with the amplifier at maximum gain, to raise the plate voltage of tube V3 above the cathode voltage. The agc circuit keeps the output of the amplifiers constant for the marking impulse, and causes this impulse to have a constant amplitude. Furthermore, the agc circuit time constants are long enough to hold the bias on the V1 tube during a no-tone or spacing condition. This is to prevent enough noise from being amplified, during the spacing interval, to operate falsely the REC relay to mark.

b. Detector. In order to be able to operate the teletypewriter receiving mechanism at the receiving station, the pulses of a-c signal shown in figure 190 must be converted into d-c pulses. This necessitates *rectifying* the signal and filtering out the carrier signal, leaving only the d-c marking pulse. Output transformer T2 couples the output of tube V2 to the full-wave copper-oxide rectifier RV1. The full-wave rectified carrier signal voltage appears across the shunt combination of R16 and C8, which filters out the carrier frequency but not the telegraph signal frequency.

c. D-c Amplifier. The d-c pulses furnished by the detector must be *amplified* in order to operate the receiving relay at the teletypewriter to the marking position. The rectified voltage appearing across resistor R16 is applied to the grid of d-c amplifier-tube V4 through resistor R17, which also prevents excessive grid-current flow in tube V4. The cathode of tube V4 is biased positive by the voltage drop across resistor R58 and 100-ohm part of resistor R59. The amplified plate current signal of tube V4 operates the polar receiving relay, REC, to its marking contact. The plate current is approximately 6 milliamperes through winding 6–3 of the relay. The biasing current is approximately 0.3 milliampere through winding 7–2 of the relay. When no carrier signal is being received during the spacing intervals, tube V4 is cut off by its positive cathode bias. The plate current drops to zero, the IR drop across R64 caused by plate current drops to zero, and allows the bias current to increase to approximately 6 milliamperes, to operate the REC relay to its spacing contact. The d-c mark and space signals then are fed to the loop circuit.

155. Complete Channel of Carrier Telegraph Using ON-OFF Modulation and Half-Duplex Loop

a. General. The operation of the overall channel is shown by transmitting a mark and a space signal from the teletypewriter connected to terminal A (fig. 191) to the teletypewriter connected to terminal B. Notice that while the loop at terminal A is neutral to negative battery, the loop at terminal B is neutral to positive battery. If one

loop is arranged for half-duplex operation, the distant loop also is arranged for half-duplex operation, but it may be polarential, neutral to negative battery, or neutral to positive battery. This is possible because each d-c loop circuit is complete.

b. Transmitting Mark Signal from Terminal A to Terminal B.

(1) When half-duplex operation is employed, the receiving terminal must be in a receiving condition (marking condition). This is necessary since the REC relay at terminal A must be operated to the mark (M) contact before the transmitting teletypewriter at A can send out copy. To transmit a mark, one of the transmitting contacts of the teletypewriter at terminal A must be closed to complete the loop circuit. The path of current is from the negative 115-volt battery through the 41–C relay, the loop wire, SL RES, the line winding (2–7) of the SEND and BK relays, L2, the mark contact of the REC relay, to positive battery. The return path of the current is through ground.

(2) A second path of current is from the negative terminal of the 130-volt battery at the mark contact of the REC relay through ground, through the artificial line (composed of resistors R69 and R67, capacitor C58, SEND BIAS RES, and resistor R68) through the bias winding of the SEND and BK relays to the positive terminal of the 130-volt battery at the mark contact of the REC relay.

(3) The current through the loop operates the 41–C relay to the mark contact which closes the circuit of the selector magnet of the teletypewriter. Current flows through the selector magnet, which causes the mechanical train of operations for a mark signal to be set up in the receiver unit. The path of current through the line windings of the SEND and BK relays is in a direction to cause their armatures to operate to the M contact because a potential is applied to the low-numbered terminal of the winding. However, the current in the bias windings is in a direction to cause the SEND and BK relays to operate to space. The reason that the two relays operate to the M contact is that the current in the line winding is adjusted to 60 ma by the SL RES while the current in the bias winding is about 30 ma. The armature of the SEND relay on the M contact causes R35 to be shorted out and allows the oscillator V5 output to go out through the sending filter, the line coils, and over the line to the B terminal.

(4) At the B terminal, the frequencies of this particular channel are passed by the receiving filter of this channel and are blocked by all the other filters. The carrier is amplified, rectified, and the resulting d-c signal then is amplified by the V4 tube. The signal causes plate current to flow in the plate circuit of the V4 tube and also through the line winding (6–3) of the REC relay, since it is part of the plate circuit. The flow of current in the line winding of the REC relay is in a direction to cause it to operate to the M contact. The armature of the REC relay applies negative potential to the loop. Current flows from the M contact of the REC relay through L2, the line winding of the SEND and BK relays, R71, the SL RES, the loop wire, the 41–C relay, the transmitter contacts, to the positive terminal of the series aiding battery, and returns through ground. The current through the 41–C relay causes it to operate to mark. This completes the circuit of the receiver magnet of the teletypewriter and sets up the mechanical train of operations in the receiving unit for a mark signal.

(5) The current flow in the line winding of the SEND and BK relays is in a direction to cause them to operate to the M contact. The current in the bias winding (7–2) is in a spacing direction. The current in the line windings, however, is about double that in the bias winding; therefore, the SEND and BK relays operate to the M contact. It is necessary for the SEND and BK relays to be on the M contact because carrier tone must be transmitted to the A terminal to hold the REC relay there to the M contact.

c. Transmitting Space Signal from Terminal A.

(1) The transmitter contacts at the teletypewriter at A are open for this interval, thus opening the loop circuit. The spring on the 41–C relay pulls the armature to space, thus opening the circuit of the receiving magnet. The mechanical train of operations for a space signal is set up in the receiving unit. Current does not flow through the line winding of the SEND and BK relays, but current does flow through the bias windings of the SEND and BK relays in a direction to cause them to operate to space. Notice that the REC relay must remain operated to the M contact to accomplish this. Carrier tone must be received from terminal B then, while signals are being transmitted from terminal A if the operation is on a half-duplex basis. When the SEND relay is operated to the S contact, the primary of T6 is shorted, and no carrier tone is transmitted to terminal B.

(2) At terminal B no tone is received; therefore, no current flows in the V4 tube or the line winding of the REC relay. The current flowing in the bias winding of the REC relay causes it to operate to the S contact. This applied positive battery to the loop. Positive battery is applied also at the other end of the loop circuit. Since there is practically no potential difference across the loop, only a small amount of current flows through the loop (3 ma). The armature of the 41–C relay will be pulled to the space contact by the spring, thus opening the receiving magnet circuit. No current flows through the receiving magnet and the mechanical train of operations is set up for a space signal.

(3) There is no current flow through the line windings of the SEND and BK relays, but there is a current flow through the bias windings. Notice that the current flow is in a direction to cause the SEND and BK relays to remain operated to the M contact because the polarity of the potential applied to the bias windings was changed when the REC relay operated to the S contact. The path of current is from the negative battery at resistor R69 as well as from ground at resistor R68 through the bias windings of the SEND and BK relays, through the S contact of the REC relay, to the mark contact of the BK relay to positive battery. The SEND and BK relays are are on the M contact, although a space signal has been received. The SEND relay must be on the M contact to short out R35 and allow the carrier tone to be transmitted to terminal A to hold the REC relay there to the M contact. The REC relay at the A terminal must be on the M contact to control the polarity of battery applied to the bias windings of the SEND and BK relays. The BK relay at terminal B must be on the M contact to apply positive potential to the S contact of the REC relay and, in turn, to the loop.

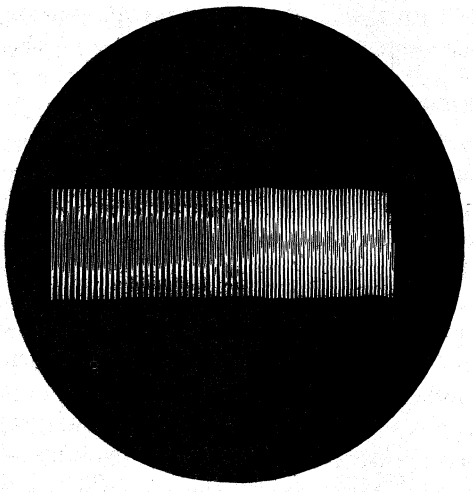
(4) Note that home copy is received when the operation is half-duplex. This is true because the transmitted signals are received by the receiving magnet of the same teletypewriter as well as that of the distant teletypewriter. Transmission from the B terminal to the A terminal is identical with the transmission from the A terminal to the B terminal.

156. Nature of Frequency-Shift Signal

In general, voice-frequency teletypewriter systems fall into two categories, those that represent the signal elements of the teletypewriter code by an on-off carrier, and those that represent the signal element by an f-s (frequency-shift) signal. In the on-off system, surges of alternating current go to the transmission line for the marking impulses, and no current goes to the line during the spacing intervals (fig. 191). In the f-s system, marking impulses are transmitted by a carrier signal of one frequency, and spacing impulses by a carrier signal of the same amplitude but of a lower frequency. The f-s signal in figure 192 illustrates both a spacing and a marking impulse. The lower frequency portion of the signal at the left is the spacing pulse. The higher frequency portion of the signal at the right is the marking pulse.

157. Elements of Frequency-Shift System

The essential components of any carrier telegraph set are an oscillator, a modulator, filters, and a detector (demodulator) (par. 150*b*). In an



TM 679-338

Figure 192. Constant-amplitude frequency shift signal.

f-s system, the modulator and the detector necessarily differ from their counterparts in an a-m system. One additional component, a limiter, usually is required in an f-s system. The operation of these components of an f-s telegraph set is explained in paragraphs 158 and 159.

158. Modulator and Oscillator

a. The essential parts of an oscillator and f-s modulator are shown in figure 193. The oscillator is of the usual type, comprising a tube acting as an amplifier, a frequency-determining network consisting of an inductance and a capacitance, and a feedback network across which is developed the positive feedback necessary to sustain the oscillations.

b. In the f-s system, the sending relay, operated by d-c signals received from the sending loop, causes a change in the frequency of the output

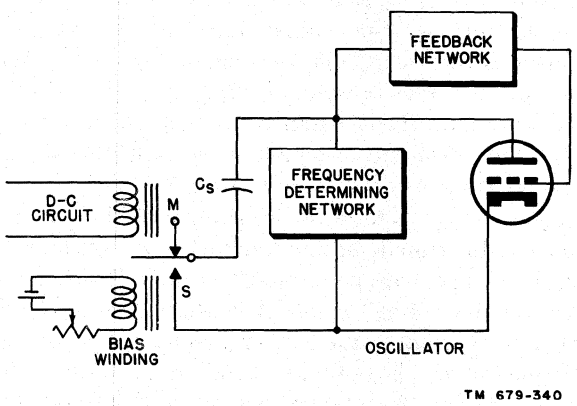


Figure 193. Essentials of frequency-shift modulator and oscillator.

signal of the oscillator. When marking pulses are being transmitted, causing the sending relay to be operated to the marking contact M, the circuit containing capacitor C_s is open, so that this capacitor is not connected across the frequency-determining network. Thus, the frequency for the marking pulse is determined only by the values of inductance and capacitance in this network. When the sending relay is operated to spacing contact S, capacitor C_s shunts the frequency-determining network. There is a resulting increase in the total capacitance in the network, causing a decrease in the frequency of the oscillator output. In this manner, the different marking and spacing frequencies are produced.

c. There are two differences between the outputs of the oscillators in on-off and f-s systems. A comparison of figures 190 and 192 shows that in the f-s system the carrier signal is of constant amplitude, whereas in the on-off system the amplitude changes. A comparison of the circuits of the modulator in figures 188 and 193 shows that the modulator in the on-off system comes after the oscillator, whereas in the f-s system it affects the oscillator. The modulator in the on-off system causes the carrier signal to be applied to the transmission line only for the marking pulses. In the f-s system, the carrier signal is always applied to the transmission line during transmission of copy.

159. Amplifier Limiter

a. The carrier signal in any f-m (frequency-modulation) system, such as the f-s system, is of constant amplitude. It is essential that the amplitude of the carrier signal applied to the f-m detector remain constant. Any variation in the amplitude is effectively the superposition of a-m upon the f-m signal. When such a signal (f-m plus a-m) is the input signal to an f-m detector (discriminator), the output of the detector is distorted. The output should be produced only by the frequency variation, but it is distorted by the undesired additional output produced by the amplitude variation. To remove this undesired variation of the amplitude of the carrier signal, if it exists, one or more *limiter* stages are included in the receiver circuit preceding the detector.

b. A limiter stage removes either or both of the extremities of the input wave to the limiter, and in this way prevents the input voltage from swinging too far in the positive or negative direction, or both. A, figure 194, shows an assumed sine-wave input to a limiter stage. If the limiter stage is

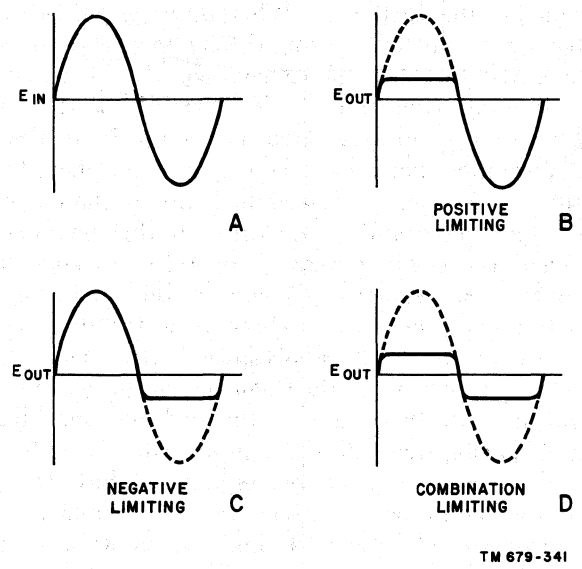


Figure 194. Output of limiter stage.

such as to cause positive limiting, as in B, the peak of the positive portion of the input is effectively prevented from swinging to the maximum positive value of the input. With negative limiting, as in C, it is the negative peak of the input that is effectively prevented from swinging to its maximum value. With combination limiting, shown in D, both the positive and negative half-cycles are prevented from swinging to their maximum values.

c. If some form of interference is causing amplitude modulation of the f-s signal, there will be an increase of the amplitude of the carrier signal in both the positive and negative directions. To limit the amplitude increases in both directions, a limiter capable of combination limiting must be used. A limiter amplifier is such a circuit. It is an amplifier so biased that the positive swings of the input signal cause the flow of grid current, and the negative swings of the input signal drive the grid voltage beyond the cut-off point of the tube. The flow of grid current during the positive portion of the input causes positive limiting of the input. The 180° phase inversion of the signal between grid and plate causes positive limiting in the grid circuit to appear as negative limiting in the plate output. The swing of the grid voltage beyond cut-off during the negative portion of the input does not produce negative limiting in the grid circuit of the amplifier limiter. But, while the amplifier tube is cut off because the grid is driven very negative, its plate voltage remains equal to the value of the applied voltage and does not vary inversely with the negative peak of the input signal. The result is positive limiting of the output.

160. Discriminator

In the f-s system, marking and spacing pulses are transmitted by signals of different frequencies. The f-m oscillator, which produces the different frequencies, is controlled by d-c pulses from the sending loop. At the receiver, the signals of different frequency (fig. 192) must be reconverted into the d-c pulses required for operation of the teletypewriter printer at the receiving station. Thus, the demodulator must perform two functions. First, it must convert the frequency-modulated carrier signal into an amplitude-modulated carrier signal; then it must rectify the amplitude-modulated carrier. The *discriminator* accomplishes both functions, delivering an output consisting of d-c pulses of different amplitude.

a. The components of a typical discriminator with its rectifier portion arranged as a voltage doubler are shown in figure 195. The primary of T1 is tuned by C1 to a frequency slightly above the mark frequency; similarly, the primary of T2 is tuned by C2 to a frequency slightly below the space frequency. The two primary windings are connected in series to the plate circuit of the preceding limiter. The output of the limiter thus appears across the two parallel tuned circuits, and is divided in proportion to their impedance. The impedance of a tuned circuit is maximum at resonance and decreases as the frequency departs from resonance. The amplitude thus also decreases as the signal frequency departs from the frequency at which the tuned circuit A or B is resonant.

b. Reference to figure 196 will aid in understanding this action of the tuned circuits in converting an f-m signal into an a-m signal. Curve S represents the resonance curve of tuned circuit A. The resonant frequency is f_a. Curve M represents the resonance curve of tuned circuit B. The resonant frequency is f_b. The space carrier frequency is f_s and the mark carrier frequency is f_m. These curves show how the voltage across the tuned circuit varies as the frequency varies. The voltage across B is point 1 on curve M and the voltage across A is point 2 on curve S. When the input is the carrier signal at the spacing frequency f_s, the voltage across A is point 3 on curve S and the voltage across B is point 4 on curve M. As the frequency of the input signal

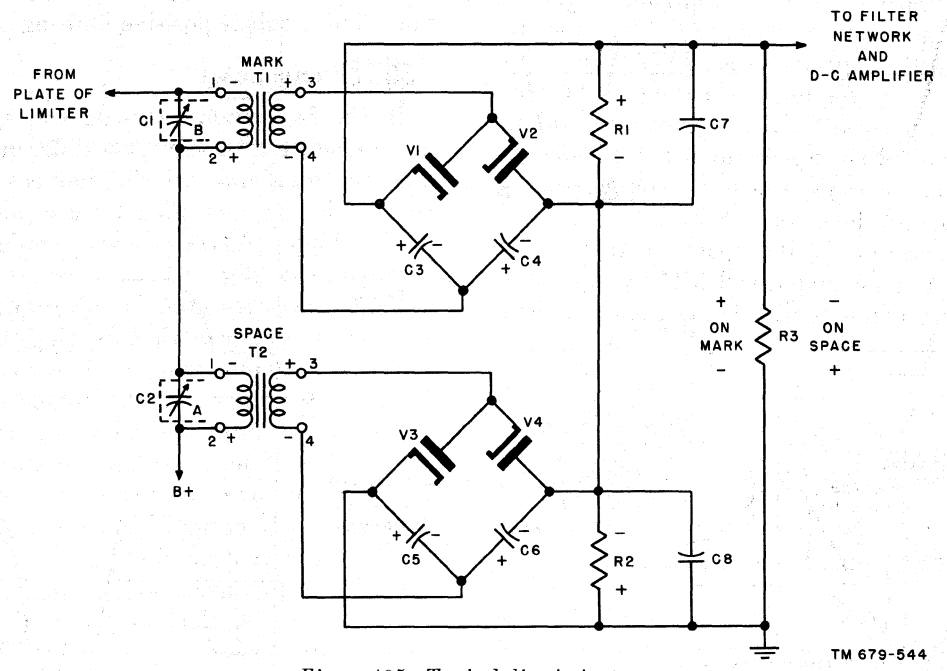


Figure 195. Typical discriminator.

goes from marking to spacing, the signal voltage across A goes from minimum to maximum, and the signal voltage across B goes from maximum to minimum. The frequency shift from marking to spacing has been converted into a change of amplitude. The signal voltage across B is shown in figure 197. Note that this signal is an a-m carrier signal. (See how if differs from the frequency-modulated carrier signal in fig. 194.) The amplitude modulated carrier signals across tuned circuits A and B then are applied to the detector-doubler circuits by transformers T1 and T2.

c. On carrier half-cycle, when terminal 3 of the transformer T1 is positive (fig. 195) and terminal 4 is negative, tube V1 conducts, because its plate is positive with respect to its cathode. The conducting of V1 charges capacitor C3; the upper plate of the capacitor is positive in relation to the lower plate, by a voltage equal to the peak voltage of the input signal. For this half-cycle, V2 is cut off, because its cathode is more positive than its plate.

d. During the next half-cycle, terminal 4 of transformer T1 is positive and the opposite ter-

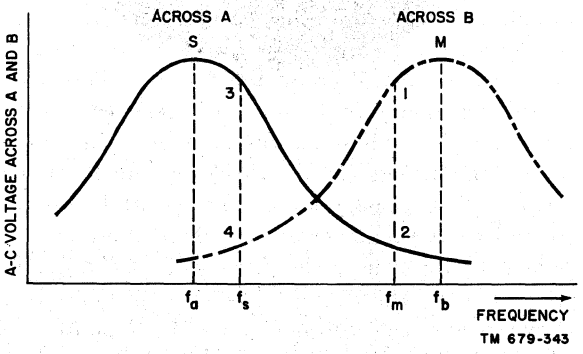


Figure 196. Resonance curves of tuned circuits in discriminator.

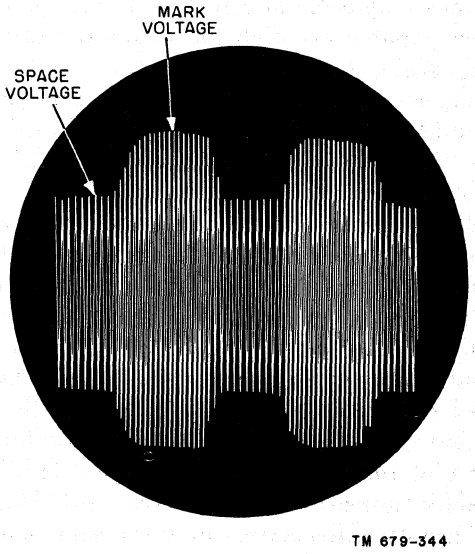


Figure 197. Voltage across tuned circuit B in linear discriminator.

minal is negative. Tube 2 conducts, and C4 rapidly charges; the lower plate of the capacitor is positive with respect to the upper plate. For this half-cycle, V1 is not conducting. The voltages across capacitors C3 and C4 are series aiding; therefore, the voltage appearing across resistor R1 is approximately twice the peak voltage that is developed across the secondary of T1.

e. The operation of the space portion of the discriminator stage is identical with the operation of the one for the mark signal that has been described.

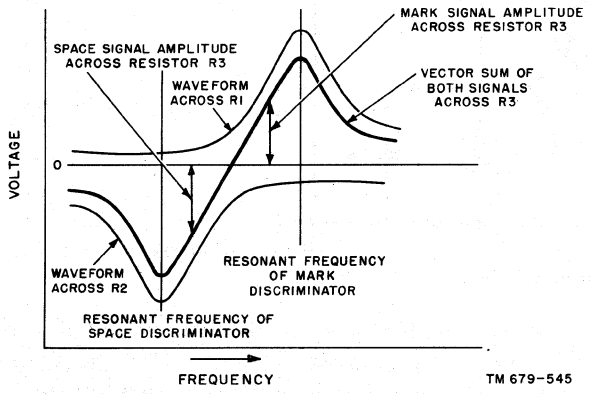


Figure 198. D-c output voltage of discriminator.

f. The voltage polarities developed across R1 and R2 are opposite in sign, as indicated in figure 198. C7 and C8 are filter capacitors. The net voltage across R3 is the vector sum of the voltages developed across R1 and R2. The voltage across R3 is therefore positive for a mark signal and negative for a space signal.

g. A, figure 199 shows the negative d-c output across R2C8. The maxima correspond to spacing pulses, the minima to marking pulses. B illustrates the positive d-c output across R1C7. The minima correspond to spacing pulses, the maxima to marking pulses. This corresponds to the amplitude-modulated carrier signal developed across tuned circuit B, which is tuned to the marking frequency, f_m. The net output voltage across R3 is the difference between the voltages across R2C8 and R1C7. This voltage is shown in C. An oscillograph of the actual output of the discriminator is shown in figure 200. This signal obviously consists of the marking pulses and spacing intervals necessary for the operation of a teletypewriter printer.

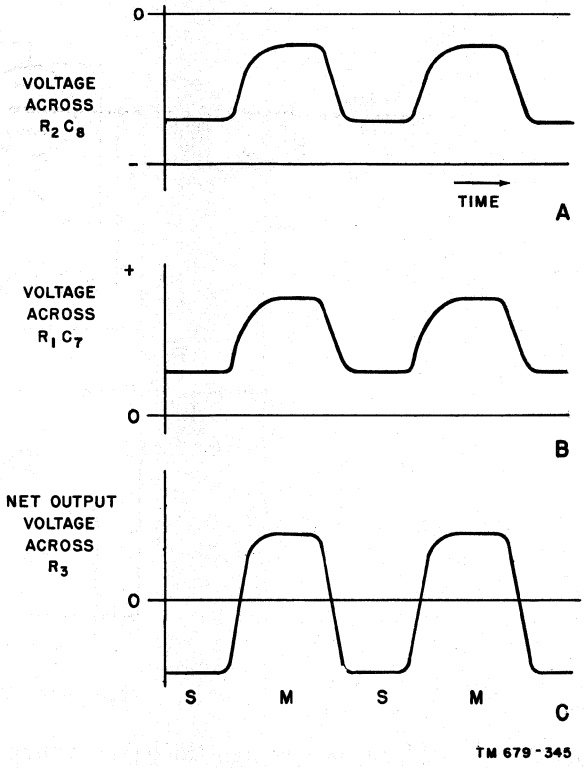


Figure 199. Rectified outputs across filters in discriminator.

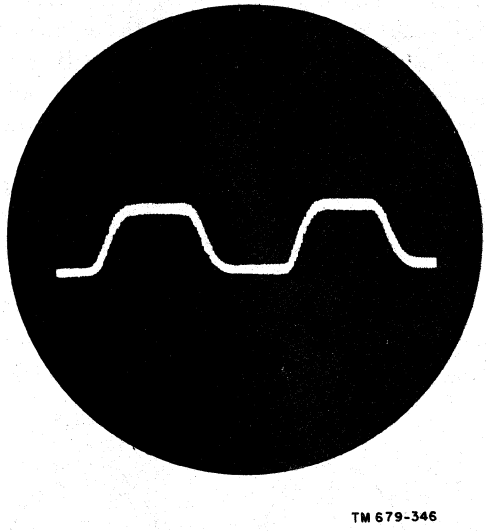


Figure 200. Rectified output of discriminator.

161. D-c Amplifier

a. The d-c output of the discriminator, consisting of marking pulses and spacing intervals which correspond to the character transmitted from the sending end, is applied to the control grid of a d-c

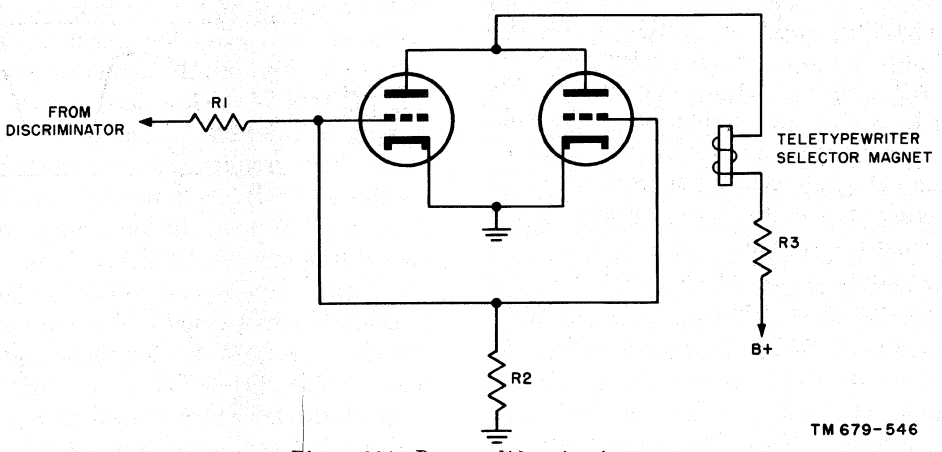


Figure 201. D-c amplifier circuit.

amplifier. The selector magnet of the teletypewriter is connected in series with the plate circuit of the d-c *amplifier*. The positive marking pulses from the discriminator cause the amplifier to conduct, and the resulting flow of plate current passes through the selector magnet. The negative spacing pulses from the discriminator drive the control-grid voltage of the amplifier beyond cutoff, and no current flows through the selector magnet. The operation of the discriminator thus controls the operation of the selector magnet.

b. The circuit of a d-c amplifier (fig. 201) consists of two tubes operated in parallel. The amplifier is operated with zero bias. During the positive half-cycles of the input to the control grid of the amplifier, the grid voltage goes positive and grid current flows. The resulting voltage drop across the grid limiting resistor R1 bucks the input signal voltage. As the input signal goes more positive, the grid current increases, as does the bucking voltage. The result is that the positive portion of the input signal is limited. The flattening-off of the grid voltage caused by positive limiting causes the plate current to be constant. During the negative half-cycles of the input to the control grid, the control-grid voltage is driven beyond the cut-off point of the tube. The plate current is zero while the tube is cut off.

c. R2 is the grid return and R3 is a current limiting resistor.

d. The no-current portion of the output of the amplifier corresponds to a spacing interval. The current portion of the output of the amplifier corresponds to a marking interval. The flattening-off of the current pulse decreases any distortion there may be of the marking pulse.

162. F-s Modulated Carrier Telegraph Channel Using D-c Loop Circuit

a. General. Figure 202 illustrates the sequence of the components in an f-s carrier telegraph channel. The block diagram shows two channels of a carrier telegraph terminal with their auxiliary circuits omitted. CHAN–1 is operating with a d-c loop, and CHAN–2 is operating with a voice frequency loop. The frequencies transmitted by the CHAN–1 oscillator are 382.5 cps for a space signal and 467.5 cps for a mark signal. The frequency during a mark interval is 42.5 cps above the channel center frequency (425 cps) and the frequency during a space interval is 42.5 cps below. The received frequencies for mark and space are 637.5 and 552.5 cps respectively in CHAN–1. The components shown by dotted lines are those in the circuit when a voice frequency loop is employed but are not energized when a d-c loop is used.

b. Sending a Mark Signal. The transmitter contacts close the loop, allowing current to flow in the loop and modulator circuit. The modulator during this interval causes the line oscillator to transmit a frequency of 467.5 cps. Capacitance is removed from the tank circuit of the oscillator, causing it to oscillate at the higher frequency of 467.5 cps. The next stage is the line output amplifier which provides an impedance match between the impedance of the oscillator and the input impedance of the filter and prevents the filter from loading down the oscillator. The 425 cps send filter passes the frequencies of this channel to the distant carrier telegraph terminal and blocks all others outside of the 382.5 to 467.5 band.

c. Transmitting a Space Signal. The transmitter contacts open the loop circuit, causing the loop current to drop to zero. The modulator circuits cause capacitance to be added to the tank circuit of the oscillator, and the oscillator transmits the lower frequency of 382.5 cps toward the line.

d. Receiving a Mark Signal. The distant carrier terminal CHAN–1 oscillator transmits signals of 637.5 cps during a marking interval and signals of 552.5 cps during a spacing interval. The 595 cps REC filter passes these frequencies and blocks all others outside of the 552.5 to 637.5 band. Signal voltages are developed across the primary of the T7 transformer and are induced across the secondary windings. These signals are coupled to the limiter stages which provide a fairly constant input voltage to the discriminator for varying signal strengths coming in from the line. The d-c output of the discriminator during a marking pulse places a positive voltage on the grid of the d-c loop output amplifier which then conducts current. The current flows through the plate circuit of the d-c amplifier the loop and the receiving selector magnet of the teletypewriter.

e. Receiving a Space Signal. During a space pulse, a signal of 552.5 cps is received from the distant carrier terminal, passed by the REC filter, the T7 transformer, the limiter stages, and applied to the discriminator. The d-c output of the discriminator during a spacing interval places a negative voltage on the grid of the d-c loop output amplifier. This negative voltage biases the d-c amplifier at cut-off. Consequently, there is no current flowing in the plate circuit of the amplifier or the loop and receiving selector magnet of the teletypewriter.

163. F-s Modulated Carrier Telegraph Channel Employing a Voice-Frequency, Frequency-Shift Loop

a. General. The block diagram of CHAN–2 of the carrier telegraph terminal of figure 202 illustrates a carrier telegraph channel using a four-wire, v-f (voice-frequency), f-s loop. The path of signals transmitted from the teletypewriter to the carrier telegraph terminal is from the transmitter contacts through the upper section of the TH–5/TG, through the upper two loop wires to the carrier telegraph terminal. The path of signals received by the selector magnet of the teletypewriter is from the carrier telegraph terminal through the lower two loop wires, through the lower section of the TH–5/TG to the teletypewriter. The loop is called a four-wire, v-f, f-s loop because four wires are used—two wires for transmission and two for reception—because the signals are in the v-f band, and the type of modulation used is f-s. This loop usually is referred to simply as a four-wire, v-f loop. The frequencies transmitted by the line oscillator of CHAN–2 are 772.5 cps during a space pulse and 807.5 cps during a mark pulse. The received frequencies of CHAN–2 and 892.5 cps during a space pulse and 977.5 cps during a mark pulse. The frequencies sent to and received from the loop are 1,225 cps for space and 1,325 cps for mark. The TH–5/TG is a single-channel telegraph terminal which transmits a 1,325-cps mark and a 1,225-cps space and receives the same frequencies. The teletypewriter is connected to the TH–5/TG. The diagram illustrates a four-wire v-f, f-s loop; however, a two wire v-f, f-s loop can be used. The components in the carrier terminal used when d-c loop operation is employed are omitted from the drawing, since these components are not energized when v-f operation is used. The auxiliary circuits are not shown.

b. Transmitting a Mark Signal. The transmitter contacts are closed, allowing current to flow in the d-c loop and through a grid resistor in the d-c amplifier. This applies a positive voltage to the grid, and the tube conducts. The signal is passed through the time delay circuits to the modulator. During a mark signal, the modulator shunts a portion of the capacitance in the tank circuit of the TH–5/TG oscillator, causing the marking frequency of 1,325 cps to be transmitted through the output amplifier, the output circuit (transformer and impedance matching components), to the v-f loop. The signals in the loop are v-f and produce signal voltages across the primary of the T1 transformer in the carrier terminal. These voltages are induced across the secondary winding and applied to the limiter stages. The output of the limiter stages is applied to the discriminator which converts the v-f mark signal to a positive d-c potential which is applied to the modulator circuit. The modulator then removes capacitance from the tank circuit of the line oscillator, causing it to send out an 807.5-cps mark through the line output amplifier and send filter over the line to the distant carrier telegraph terminal.

c. Sending a Space Signal. The transmitting contacts of the teletypewriter open, and prevent current from flowing through the grid resistor of the d-c amplifier. The tube is biased at cutoff.

The space pulse is passed through the time delay circuits to the modulator which, during the space interval, adds capacitors to the tank circuit of the oscillator. The oscillator transmits a 1,225-cps space through the output circuits, through the v-f loop to the carrier telegraph terminal. The signal passes through the T1 transformer and limiter to the discriminator, which converts the v-f signal to a d-c negative voltage. This negative voltage is applied to the modulator circuit which then adds capacitance to the tank circuit of the line oscillator. The 772.5-cps space then is transmitted through the line output amplifier and the send filter over the line to the distant carrier telegraph terminal.

d. Reception of Signals from Line.

(1) *General.* Three main steps are involved in this process. The first is the conversion of the incoming f-s signals to d-c signals at the output of the discriminator; second is the conversion of the d-c pulses appearing at the output of the discriminator into v-f loop output signals; third is the conversion of the v-f loop signals into d-c pulses for reception by the receiving selector magnet of the teletypewriter.

(2) *Receiving a mark signal.* A signal of 997.5-cps is received during a marking interval, passed by the receive filter, the T7 transformer, and limiter stages, and appears as a d-c voltage at the output of the discriminator. This voltage is applied to the modulator stage which shunts capacitors in the tank circuit of the loop oscillator, causing a frequency of 1,325 cps to be transmitted to the v-f loop. The 1,325-cps signal is received by the TH–5/TG, where it appears as a positive d-c voltage at the output of the discriminator. The voltage is doubled in value by the detector doubler and applied to the grid of the d-c amplifier. During the marking interval, current flows through the plate circuit of the d-c amplifier and through the receiving selector magnet of the teletypewriter.

(3) *Receiving a space signal.* The 892.5-cps space signal is converted to a d-c voltage of opposite polarity to the voltage output when a mark signal is received by the discriminator and is applied to the modulator stage. The modulator stage adds capacitors to the tank circuit of the loop oscillator which now sends out a 1,225-cps space signal. This space signal is transmitted through the loop output stage, the T4 transformer to the v-f loop. The signal is received by the TH–5/TG input circuit, passes through the limiter stages, and is converted to d-c voltage by the discriminator. The output of this discriminator is negative during a spacing pulse. This negative voltage is doubled by the detector doubler and is applied to the grid of the d-c amplifier. The d-c amplifier is biased at cut-off by this negative voltage; no current flows in the plate circuit of the tube or through the receiving selector magnet of the teletypewriter.

164. Two-Source F-s Sending Circuit

a. General. Figure 203 illustrates the sending branch of a carrier telegraph terminal employing the two-source method of frequency-shift modulation. The sending circuit is similar to the circuit used in the ON-OFF method of modulation. The principal difference is that two such circuits are required for each channel of communications. One oscillator (425 cps) output is transmitted to the line during the marking interval and the second oscillator (595 cps) output is transmitted to the line during the spacing intervals.

b. Transmitting a Mark Signal. When the teletypewriter transmits a mark signal, the loop current of 60 ma, flowing through the M SEND and S SEND relays causes them to operate to the M contacts. The current flow in the bias windings is about 30 ma in a direction to cause the relays to operate to the S contact. The armature of the M SEND relay on the M contact shunts the 200 K resistor in the mark sending circuit and allows the mark oscillator output to be transmitted through the M1 SEND filter to the line at the proper marking level. The armature of the S SEND relay on the M contact shorts the 1–2 winding of the T6 transformer in the space channel and prevents the space carrier tone from being transmitted to the line.

c. Transmitting a Space Signal. During a spacing interval, no current flows through the loop or through the 2–7 winding of the M SEND or S SEND relays. The current flowing through the 3–6 windings causes the relays to operate to

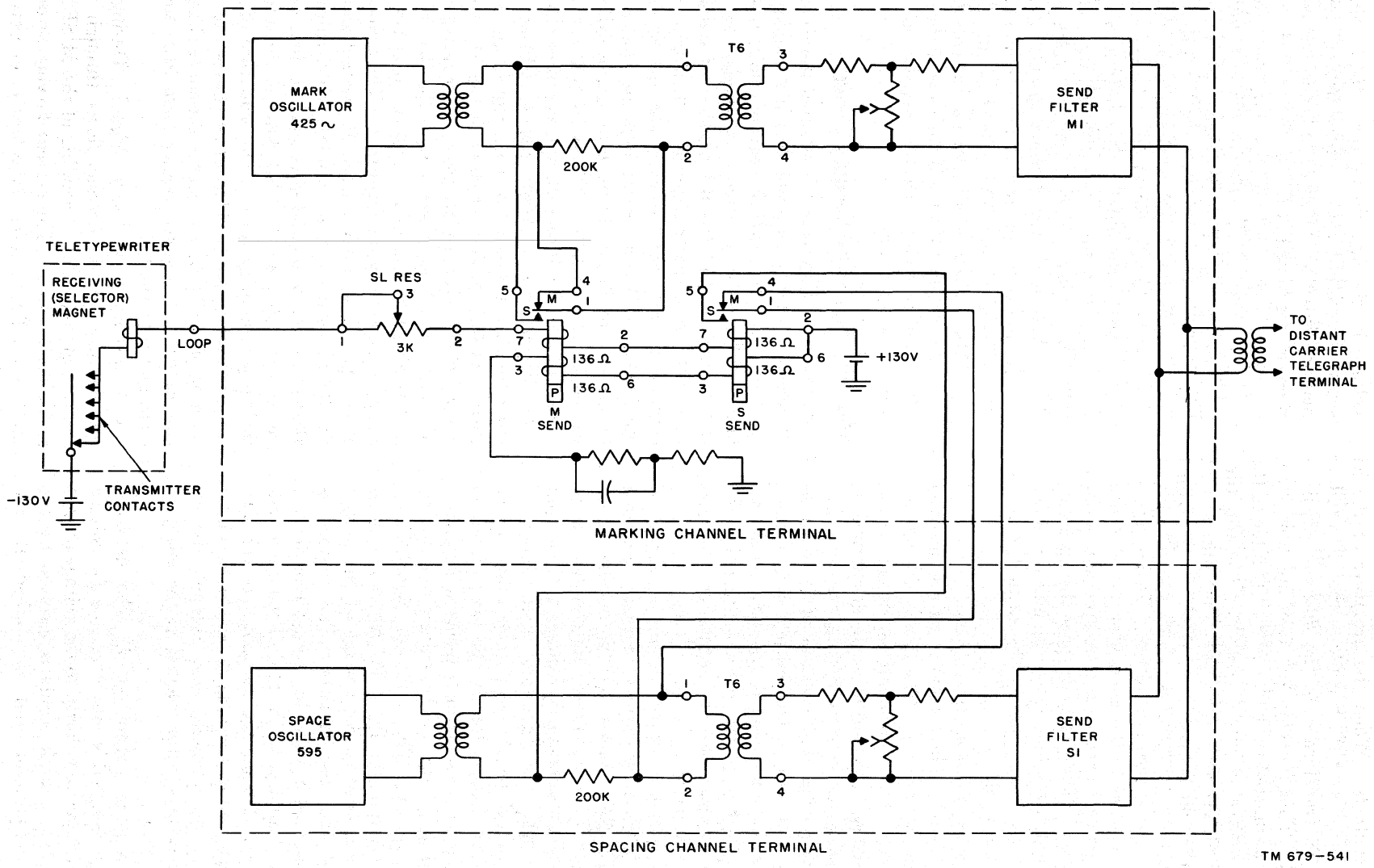


Figure 203. Marking and spacing channel terminal send circuit, schematic.

the S contacts. When the armature of the M SEND relay is on the S contact, the 1–2 winding of the T6 transformer in the marking circuit is shorted. This prevents the marking tone from being transmitted through the SEND filter to the distant carrier telegraph terminal. The armature of the S SEND relay on the S contact puts a short across the 200 K resistor in the spacing channel and allows the spacing tone to be transmitted through the S1 SEND filter to the line at the proper level.

d. Transitions. During a transition, the 200 K resistors in both sending circuits reduce the output by 6 db. This occurs because the armatures of the relays do not touch either contact. At this output level, the receiving relay at the distant terminal will not be operated. Transition delay causes bias distortion.

165. Two-Source F-S Receiving Circuit

a. General. Figure 204 illustrates the receiving branch of a channel using the two-source f-s system. Channels 2 and 3 are drawn in block diagram form and are identical with channel 1. The circuits are similar to those in which the on-off type of modulation is used—the principal difference being that there are two such circuits per channel. The two circuits terminate in a single REC polar relay which relays the d-c signals to a receiving teletypewriter. The circuits of the CHAN–1 d-c amplifiers are drawn schematically to show how the outputs of the mark and space tubes operate the REC polar relay.

b. Receiving a Mark Signal. During a marking interval, a signal of 425 cycles is received. It is passed by the REC M filter of channel 1, is amplified by two stages, and rectified by the detector stage. The output of the detector is applied to the grid of the mark d-c amplifier tube, causing plate current to flow through resistor N, the 5–8 and 3–6 windings of the REC polar relay, through the top half of resistor M, to the B+ plate battery. Resistor N and capacitor K form a filter to remove any carrier frequencies in the plate circuit. Resistor M is across a test jack not shown in the diagram. The current flow is in a direction to operate the REC relay to the M (No. 13) contact. The S resistor limits the grid current when a high level signal is received and also prevents the grid from becoming highly positive. The suppressor grid is connected to the bias potentiometer. Current flows in this circuit when a mark signal is received and develops a voltage in the agc circuit (par. 154). Since no signal is received by the space branch during a marking interval, the space tube does not conduct because it is biased at cutoff.

c. Receiving a Space Signal. During a spacing interval, a signal of 595 cycles is received and passed by the REC S filter. The signal is amplified by two stages and rectified by the detector stage. The output of the detector is impressed upon the grid of the space d-c amplifier, causing a flow of plate current through the N resistor, through 2–1 and 4–7 windings of the REC polar relay, through the lower half of the M resistor to the B+ battery. The current through these windings is in a direction to cause the armature to operate to the S (No. 15) contact. The mark tube is biased at cutoff during the spacing interval and does not draw current.

d. Transmitting to Loop. When the REC polar relay is on the M contact, negative battery is applied to the loop and a current of about 30 ma flows from the M contact of the REC relay through the RL loop, the 6–3 winding of the line polar relay, and returns through ground. The polar relay operates to the mark contact, closing the receiving selector magnet circuit. When the REC polar relay is on the S contact, current flows from the negative terminal of the battery, through ground, the 6–3 winding of the line polar relay, through the RL loop to positive battery. The line polar relay is operated to the S contact, opening the receiving selector magnet circuit. The signals sent over the RL loop to the teletypewriter are polar—that is, reversals of current flow.

e. 209–FG Polar Relay. A, in figure 205, shows the number of windings on the 209–FG polar relay used in figure 204. There are 6 windings—2 of these, the 10–12 and the 9–11, are used as the bias windings, and the other 4 are used as the line windings. The resistance of the line and bias windings differ. The line windings have 185 ohms resistance whereas the bias windings have a resistance of 120 ohms. B, figure 205 shows the direction of electron current flow through the windings to cause the relays to operate to the M (No. 13) contact. Notice that, for the 7–4 and 11–9 windings, current flowing in the direction from 7 to 4 or 11 to 9 has the same effect as current flowing from 3 to 6 in the 3–6 winding. Furthermore, the rule given in paragraph 128 for determining the operation of the 215 or 255 relay armature to the mark or space contact by

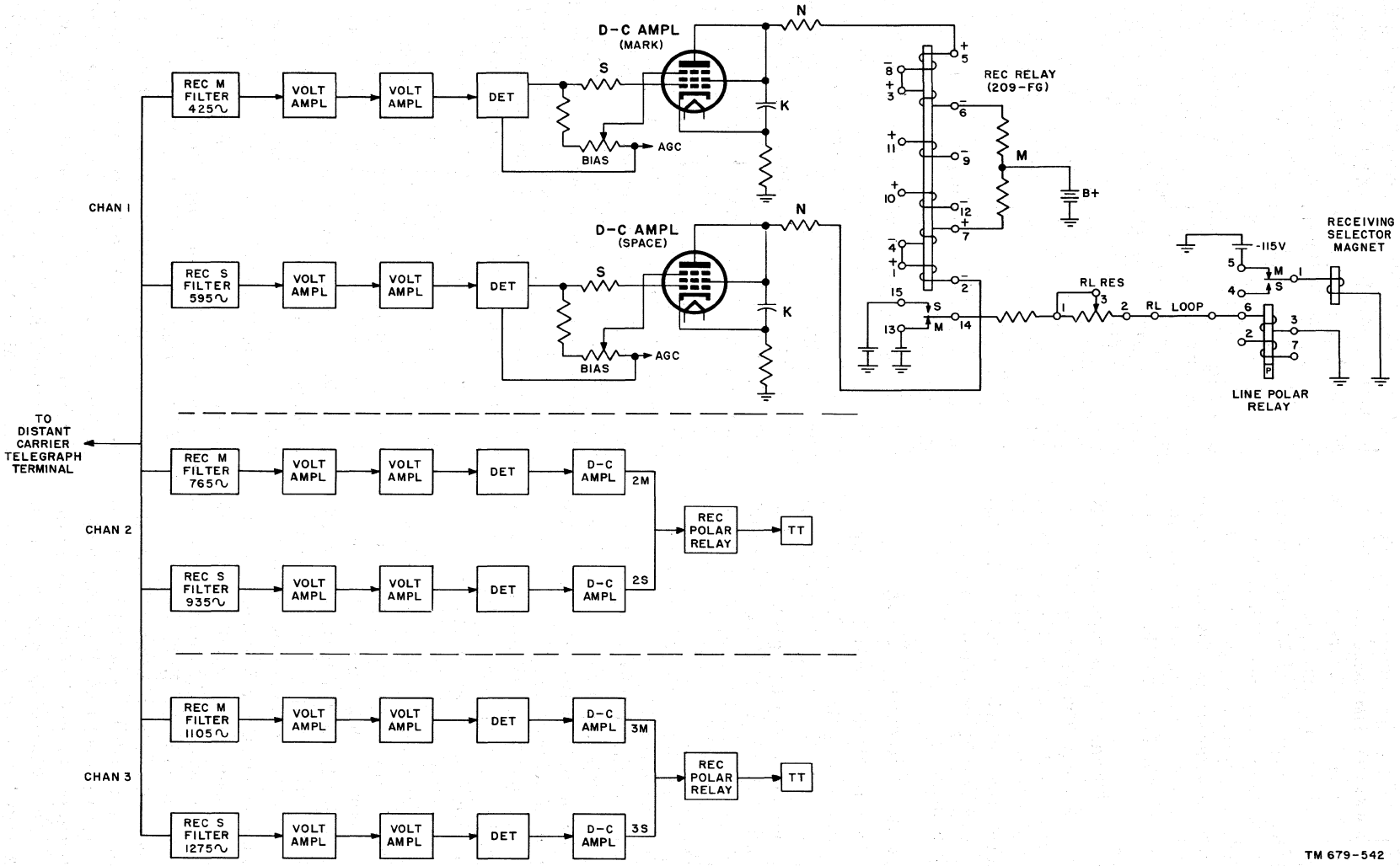


Figure 204. Two-source, F-s receiving circuit.

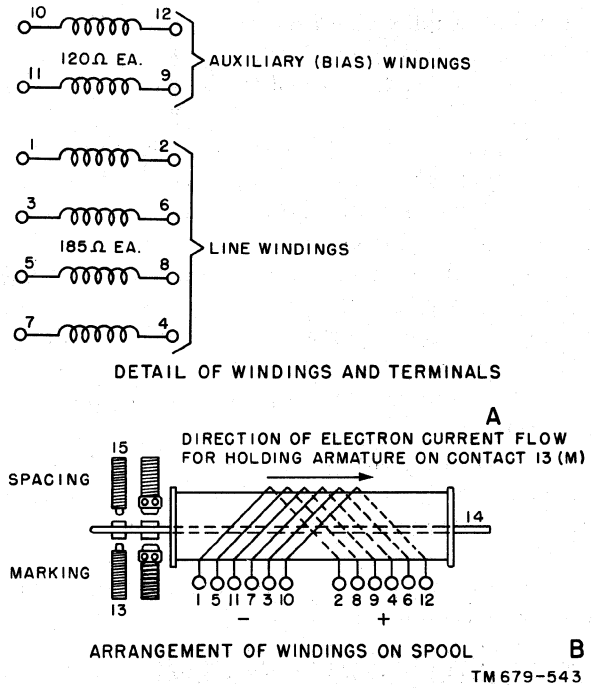


Figure 205. 209–FG polar relay.

referring to the applied potentials is exactly the reverse, except for the 4–7 and 9–11 windings.

166. Summary of Elements of Carrier Telegraph Systems

a. Carrier Telegraphy. Carrier telegraphy makes it possible to transmit several messages at the same time, and increases the distance over which messages can be transmitted. The different methods of carrier telegraph operation are the on-off method, the single-sideband method, the frequency-shift method, the two-source method, and the one-source two-band method.

b. Elements of On-Off System. In the on-off system, the oscillator produces the carrier frequency. Two frequencies are assigned to each channel; consequently, each channel oscillator is adjusted to send either of the two frequencies assigned to the channel. The modulator is controlled by d-c pulses from the sending loop. It causes the oscillator to be connected to the transmission line for marking pulses, and to be short-circuited during the spacing intervals. The sending filter suppresses all sideband frequencies which are not necessary for satisfactory transmission of signals, and which might interfere with adjacent channels. The receiving filters guide the carriers and sidebands into the proper receiving circuits. An amplifier increases the amplitude of the carrier signal. The agc circuit causes the carrier signal to have a fairly constant amplitude. The detector converts the pulses of carrier signal into the d-c pulses necessary to operate the receiving relay.

c. Elements of Frequency-Shift System. In the f-s system, the oscillator produces the carrier frequency. The modulator connects a capacitor across the L–C circuit of the oscillator during the spacing intervals. During the marking intervals, this capacitor is not connected. Therefore, the spacing frequency is less than the marking frequency. The amplifier limiter prevents changes in the amplitude of the carrier. The discriminator with its associated detector-doubler, converts the f-s carrier signal into an a-m carrier signal, and then produces the d-c pulses necessary for operation of the receiving relay. The d-c amplifier is controlled by the d-c pulses from the discriminator, and causes a current to flow in the receiving selector magnet of the teletypewriter for the marking pulses.

d. Elements of Two-Source Frequency-Shift System. Two oscillators are used to produce the carrier frequencies. The output of one oscillator is transmitted during a marking interval and the output of the second oscillator is transmitted during a spacing interval. The frequencies which the two oscillators produce are different; effectively, then, frequency-shift signals are produced. The receiving circuits are similar to those used in the a-m on-off systems, but two such circuits are necessary for each channel. The output circuits of both receiving circuits are connected to a common receiving relay. The output of the mark receiving circuit causes the receiving relay to operate to the mark contact, whereas the output of the space receiving circuit causes the receiving relay to operate to the space contact. The receiving relay transmits d-c signals to the teletypewriter.

167. Review Questions

a. List the five methods of modulation used in carrier telegraph systems. Name three components used in the f-s systems which are not found in the on-off systems.

b. What is the purpose of the REC and SEND filters in the carrier telegraph terminal?

c. Why are signals in the f-s modulated system less liable to distortion than those of the a-m modulated system?

d. What advantage does the single-sideband method have over the system which uses two sidebands?

e. Why is a different frequency used for each direction of transmission? What would happen if the same frequency were transmitted in both directions?

f. What is the purpose of the SEND relay in the systems employing on-off modulation? Is a SEND relay always used in carrier telegraph systems?

g. Why is an automatic gain control circuit incorporated in the receiving circuit of a carrier telegraph terminal employing on-off modulation?

h. Explain how the REC relay in the a-m system described is operated from mark-to-space and from space-to-mark by the output of the detector.

i. Why must the armature of the REC relay in the local carrier terminal be on the mark contact when half-duplex operation is used and the local teletypewriter is transmitting?

j. What is the purpose of the limiter stage in an f-s system?

k. What is the purpose of the discriminator in the f-s system? Explain how the polarity of the discriminator output changes when there is a mark-to-space transition.

l. Trace the path of a signal from a transmitting teletypewriter to the receiving teletypewriter through a channel of an a-m, on-off modulated carrier telegraph.

m. Name some advantages of a v-f loop over a d-c loop.

n. Why is it necessary to use a piece of equipment such as a TH–5/TG when the carrier telegraph terminal is operating on a four-wire, v-f basis?

o. In the two-source, f-s system described, why are two receiving circuits used?

Section V. CHARACTERISTICS OF CARRIER-TELEGRAPH SYSTEMS

168. Transmission Objectives

The objectives of setting up a carrier telegraph system are—

a. It should be possible to maintain serviceable communication at all times. This can be achieved by careful and diligent maintenance work.

b. The size and weight of the equipment should be limited. This depends only to a small extent on the electrical characteristics of the system.

c. The speed of sending should not be too slow. There should be a maximum number of channels, and a minimum of interference. These objectives can be achieved only by providing the system with the proper electrical characteristics.

169. Speed Versus Bandwidth Considerations

a. Bandwidth Versus Buildup Time.

(1) The *buildup time* of a pulse, such as a marking pulse, is the time required for the edge of the pulse to go from the minimum to the maximum electrical value of the pulse. In the series of d-c pulses shown in figure 200, the buildup time is the time required for the transition from the minimum spacing value to the maximum marking value. For the buildup time, T, to remain unchanged when such a pulse is passed through an amplifier or a filter, the bandwidth of the amplifier and filter must be wide enough for the high-frequency cutoff f_c to be equal to the reciprocal of the build-up time of the pulse, or $f_c=1/T$. This means that when a relatively long buildup time can be tolerated, the bandwidths can be relatively narrow and, when a short buildup time is required, the bandwidths must be relatively quite broad. For the transmission of pulses with instantaneous transitions, the amplifiers and filters should have infinitely broad bandwidths. If the bandwidth is too narrow for transmitting the pulses, without changing the buildup time, the result is a delay in the buildup of the pulse amplitude—that is, the buildup time is increased. The same conclusions apply to the builddown time of the transition from the marking to the spacing conditions.

(2) Although instantaneous transitions from spacing to marking amplitude, and from spacing to marking frequency, are desirable, it is highly impractical to use filters having the infinitely broad bandwidth required for the transmission of such transitions. Filters are included in carrier telegraph systems to separate the various channels in the voice-frequency range. The filters must be of relatively narrow bandwidths in order to suppress all the sideband frequencies which are not necessary for the satisfactory trans-

mission of signals in each channel, and which can cause interference with the signals in adjacent channels. There is a delay in the signal transition as a result of the use of narrow band filters, and some distortion of the transmitted signal. But the filters can be used, for a certain amount of distortion can be tolerated. There is a limit, however, on how narrow the bandwidths can be made. The narrower the bandwidth, the greater the delay in the buildup of the marking-pulse amplitude, or in the transition from spacing to marking frequency. Too great a delay causes the signal to be distorted to an extent that cannot be tolerated. The distortion resulting from the use of either too narrow a channel or too narrow a filter is shown in figure 206. A comparison of the amplitude-modulated signal in figure 198 and the distorted signal in A, figure 206, shows the manner in which the transition from the spacing to the marking amplitude has been delayed. The importance of this delay is seen in B, figure 206. This d-c signal, by means of which the teletypewriter printer is to be operated, is of the type that causes characteristic distortion.

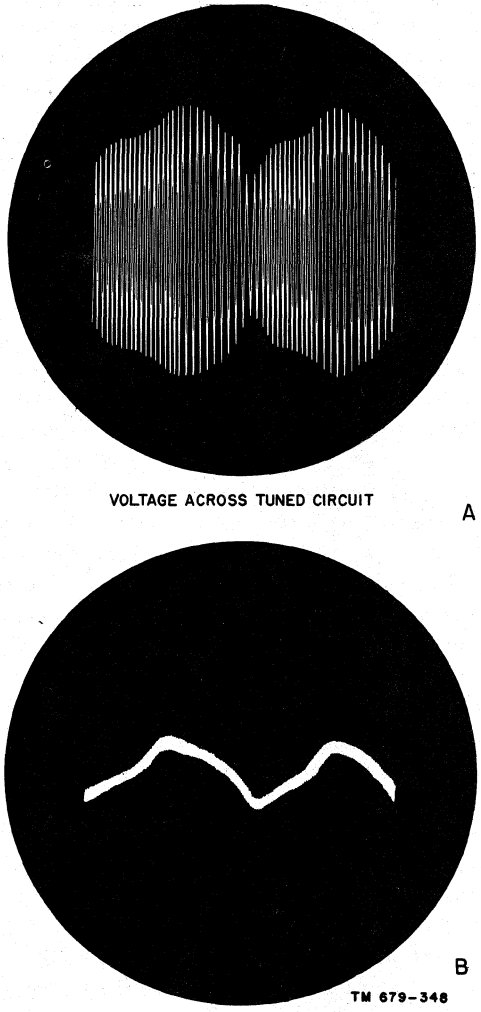


Figure 206. Effect of too narrow channel filters on waveforms in linear discriminator.

(3) These limitations on filter bandwidths apply to the filters used in both a-m and f-s carrier systems. The same types of filters are used in the two systems.

b. Sending Speed Versus Bandwidth.

(1) The relationship between the channel bandwidth and signal distortion sets a limit on the rate at which messages can be transmitted. The following numerical example illustrates this. If the channels in a carrier telegraph system are assigned bandwidths of 90 cycles, the longest buildup time of signals in the filters of these channels is $\frac{1}{90}$ second equals 11 milliseconds. At a sending speed of 100 wpm (words per minute) the duration of the signal elements making up the code symbols is 13 milliseconds. If the channel and filter bandwidths were made any narrower, the buildup time of the signals might be greater than 13 milliseconds. Space-to-mark transition of the carrier signal would not be complete before the mark-to-space transition at the end of the marking pulse occurred, resulting in distortion of the transmitted message.

(2) Thus, with a sending speed of 100 wpm, the channels and filters cannot have bandwidths of less than 90 cycles. And if the bandwidth of each channel is 90 cycles, the sending speed cannot be greater than 100 wpm. A decrease in the channel bandwidths would require a decrease in the sending speed, if serious distortion is to be avoided. An increase in the sending speed would require an increase in the channel and filter bandwidths.

(3) This conclusion can be stated in a different manner by means of the graph in figure 207. The graph shows the rela-

tionship between the distortion of the d-c signals and a factor K, which is seen to be the ratio of the channel bandwidth and twice the *dotting* speed. (The dotting speed is defined as the number of signal elements per second, divided by 2.) For values of K in the neighborhood of 1, the distortion is so severe as to cause intelligible transmission to fail completely. Values of K in the neighborhood of 5, which result in transmission with negligible distortion, can be achieved by the use of broad bandwidths; but this would limit the number of channels. The value of K must be a compromise value, which would yield a maximum number of channels and a minimum amount of distortion. For operation over long periods of time, the value of K should be about 1.4 or 1.5. The channels of the multichannel carrier telegraph terminal AN/TCC–4 are 115 cycles wide. The maximum dotting speed for K equals 1.5 would then be 115/3 equals 38 dot cycles per second. A dot cycle is composed of a closure and an open, each of equal duration. This is equivalent to a mark and a space signal or two signal elements. Thirty-eight dot cycles per second is equivalent to slightly more than 100 wpm, and our calculation demonstrates that this is the top sending speed that the system is capable of. The method of converting dot cycles per second into words per minute is as follows:

38 dot cycles per sec$\times$2=76 signal elements per sec.

1 word=5 letters and a repeated space or 6 characters.

1 character=7.4 signal elements.

1 word=7.4 signal elements$\times$6 characters=44.4 signal elements.

$$38 \text{ dot cycles per sec} = \frac{76 \text{ signal elements per sec} \times 60 \text{ sec}}{44.4 \text{ signal elements}} = 102.7 \text{ wpm.}$$

c. Relationship of Sending Speed and Line Characteristics. The characteristics of the transmission line over which the carrier signal is being transmitted also has a part in determining the maximum rate at which a message can be transmitted. The attenuation characteristics of a transmission line cause the line to act as a filter. Since the signal attenuation increases with frequency, the line attenuation may become so great as to cause effective filtering of all signals having frequencies beyond some particular cut-off value. This would tend to narrow the bandwidths of channels, particularly those in the higher frequency ranges. The sending speed would have to be reduced, to permit transmission without undue distortion over the channels, the bandwidths of which have been reduced by the filter characteristics of the transmission line.

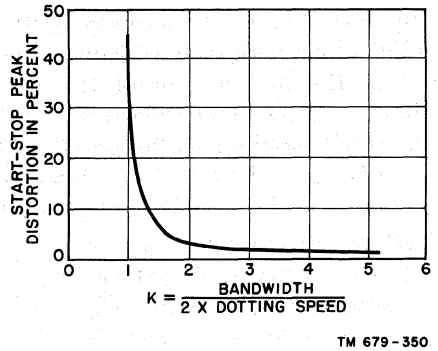


Figure 207. Distortion versus bandwidth characteristics for frequency-shift and on-off systems.

170. Maximum Number of Channels Obtainable

a. Frequency-Attenuation Characteristic. Of the transmission objectives listed in paragraph 168, the sending speed and number of channels might be classified as performance characteristics of a carrier telegraph system. A factor in determining the maximum number of channels which can be used on a given facility is the pass band of the facility. The number of channels possible in the voice range is determined by the pass band of this range. The attenuation characteristic of the filter that limits this range may also be a factor in determining the number of channels. Frequently, the cutoffs of this filter are not abrupt, nor is there constant attenuation of the signals having frequencies in the range between the cutoff points of the filter. If, as a result, the two sidebands of an a-m channel are unequally attenuated, distortion will be introduced. Similarly, in frequency-shift transmission, if the signals at the marking and spacing frequencies are unequally attenuated, amplitude modulation is superimposed on the frequency shift, and distortion results.

b. Envelope-Delay Distortion. Envelope-delay distortion usually tends to appear in the telegraph channels located near the limits of the v-f range, particularly if there are telephone filters to limit the range. It is the result of the phase distortion

caused by the filter in the vicinity of its cutoff frequencies. The permissible channels must be restricted to the mid-portion of the v-f range. If the channels are extended to the limits of the range, the resulting phase distortion of the signals in these channels might render them unrecognizable.

c. Minimum Carrier Frequency.

(1) In carrier telegraphy, the marking signals consist of pulses made up of a number of cycles of the carrier signal. As the carrier frequency is reduced (or the signaling frequency increased) the number of carrier cycles per signal element decreases. If half-wave rectification is used in the receiver, one signal element may differ from another signal element by as much as one carrier cycle, depending on the phase of the initial carrier cycle that enters the detector. If there are very few carrier cycles per signal element, as a result of the decrease of the carrier frequency, a variation in length of 1 carrier cycle will result in considerable distortion.

(2) Even when full-wave rectification is used in the detector, there may be appreciable distortion as a result of the decrease in the carrier frequency. There is no synchronization between the modulating signal and the carrier when succeeding signal elements differ in length by some part of one carrier cycle. As a result, the sending relay starts signal elements in a fortuitous (accidental) manner, with respect to the carrier cycle. The receiving relay will also start the signal elements in a fortuitous manner with respect to the carrier cycle. Signal elements may differ in length by as much as 1 carrier cycle when there is a variation in the phase of the initial carrier cycle that enters the detector. Thus, there may be considerable distortion if there are few carrier cycles per signal element as a result of using too low a carrier frequency, whether half-wave or full-wave rectification is used in the receiver.

(3) In general, it is necessary in both a-m and f-s carrier-telegraph systems that there be at least 10 or 12 complete carrier cycles per signal element when full-wave rectification is used in the receiver, and twice that number when half-wave rectification is used. If there are to be 10 carrier cycles in a signal element of 22 milliseconds, each carrier cycle has a period of 2.2 milliseconds. The carrier frequency is 455 cycles. Thus, the minimum carrier frequency that can be used with signal elements 22 milliseconds in duration and full-wave rectification, if there are to be at least 10 carrier cycles per signal element, is 455 cycles. In actual practice, the minimum carrier frequency used is 425 cycles (par. 174).

d. Maximum Carrier Frequency. The maximum carrier frequency that can be utilized is limited by the cutoff point of the filters that limit the voice-frequency range. The signals transmitted in a channel located too close to the cutoff point of the v-f filter would suffer serious delay distortion because of the nonlinearity of the phase-shift characteristic of the filter (*a* and *b* above). Furthermore, the distortion would be increased by the unequal attenuation of the upper and lower sidebands. As a result, the maximum carrier frequency is somewhat less than the upper limit of the v-f range. In a typical carrier telegraph set, the maximum carrier frequency is 1,995 cycles, which is much less than the upper limit of the v-f range—2,700 cycles. The present trend is to increase slightly both of these figures.

171. Interference

a. Design Considerations. The third transmission objective in a carrier telegraph system is that the effects of interference should be kept to a minimum. This requires that the system be so designed that the distortion resulting from the characteristics of the system is as small as possible. Interference adds, in a random manner, to the other sources of signal distortion. If the effects of the distortion are kept to a minimum, then the added distortion caused by the random interference will not cause the total distortion to be excessive. In other words, the design of a carrier telegraph system must provide a margin of safety against the distortion caused by random interference.

b. Sources of Interference. The rate at which printing errors occur in a teletypewriter system depends on the margin of safety, the nature of the interference, and the frequency with which the interference occurs. The sources of interference are interaction with other services; noise; inter-

channel interference; frequency drift; and level variations.

172. Interaction with Other Services

The interference resulting from the interaction between different services such as telegraph and power transmission and telegraph and telephone transmission, usually arises from the fact that the frequencies at which the services operate are too close to one another, or that the power level of one service is so much greater than the other. As a result of the carrier frequencies being too close to one another, the sideband frequencies developed in one service may overlap into the frequency range of the second service. For the effects of the difference in the power levels of a communication-transmission system and a power-transmission system, see chapter 2. Coordination of the operating characteristics of the various services is necessary to keep interference caused by interaction between services at a minimum. First, the frequency ranges of the various facilities must not be too close to one another. Second, there should be coordination of the relative power levels of the various services. Furthermore, all services should use the minimum current required for operation of the service. In services using f-s systems, the frequency shifts should have the minimum values compatible with efficient operation of the service. In other words, the operating characteristics of any system should be such as to make for most effective operation of the system, and yet cause the least interference with other systems.

173. Effect of Noise

a. Effect of Noise on A-m Systems.

(1) A *noise signal* has two characteristics that determine how much interference the noise signal superimposes upon a carrier signal—one is the amplitude of the noise signal; the other is its frequency, or the difference between the frequencies of the noise and carrier signals. The two effects of a noise signal on a carrier signal are to change the amplitude of the carrier and to change its frequency.

(2) With a-m carrier-telegraph systems, the change of frequency resulting from the noise signal can be disregarded, because its effect on the transmitted signal is negligible. However, the change of amplitude caused by the noise signal cannot be disregarded. It appears as a signal in the receiver. If the noise signal occurs during a marking interval, the change of amplitude distorts the marking pulse. If the noise signal occurs during a spacing interval, it may produce a false signal element in the receiver and cause a printing error.

(3) Both with random and impulse noises, the effects of the interference are rather constant at any frequency in the entire audio range. The total change of amplitude is the sum of the changes produced by all the noise signals in the audio range. However, not all these signals need to be considered in determining the total change of amplitude of a signal in one of the carrier telegraph channels. The filters which suppress all the unnecessary sideband frequencies and limit the width of the channels simultaneously suppress all the noise signals that have frequencies outside the limits of the channels. It can be seen from this that channel bandwidths have compromise values. A broad bandwidth is desirable to limit distortion; however, the result of a broad bandwidth is an increase of distortion caused by noise signals; therefore, the width of the channel must be such as to keep the total distortion from being excessive.

b. Effect of Noise on F-s Systems.

(1) Of the two changes that a noise signal causes in an f-s carrier signal, neither can be disregarded. The change of amplitude is rendered negligible, however, by the use of a limiter stage in the receiver. This stage eliminates changes in the amplitude of the carrier signal. If such changes were not eliminated, the change in the amplitude of the carrier signal resulting from interference would cause distortion of the signal elements.

(2) The change of carrier frequency caused by the interfering noise signal is superimposed upon the change of carrier frequency caused by the modulating d-c pulses from the teletypewriter sender. As a result, there is distortion of the d-c pulses from the discriminator to the teletypewriter printer. An example of the type of distortion a change of frequency can produce follows. During a spacing interval, the receiver is operating at the

spacing frequency. If a noise signal at or near the marking frequency develops at the receiver, there will be at the discriminator a transition from the spacing to the marking value of the d-c output. The space-to-mark transition at the receiver has been advanced. If the transition is advanced sufficiently, the spacing interval will appear to the selecting mechanism of the printer as a marking pulse, and a printing error will result.

(3) In f-m carrier telegraph systems, the effects of random and impulse noises are similar, so that no real distinction need be made between them. Impulse noises cause a somewhat greater amount of distortion. For both types of noise, the effects of the noise increase as the difference between the frequencies of the noise and carrier signals increases. The total effect is the sum of the effects produced by all the noise signals in the audio range. Not all these signals affect the frequency shift signal, however, for the channel filters suppress all the noise signals that have frequencies outside the limits of the channel. In an f-s system, the use of broad-band channels causes an increase in the interference that develops in the channels. As the bandwidths increase, the amount of increase in the interference increases. However, for low noise levels, broadband channels and greater frequency shift give better performance. Another point to note is that a noise signal causes less distortion to develop in a channel of an f-s system than in the channel of an a-m system, when the channels are of the same bandwidth.

c. A-m Versus F-s Systems. A comparison of the effects of noise signals on the carriers in a-m and f-s systems is shown in figure 208. The upper set of graphs is for channels of relatively narrow bandwidths—115 cycles; the lower set is for channels of broad bandwidths—740 cycles. The sending speed in both channels is 23 dps (dot cycles per second), which corresponds to 60 words per minute. Both graphs show that as the amplitude of the noise signal increases, there is an increase in the peak distortion of both the a-m and f-s signals. However, the a-m distortion always is greater than the f-s distortion. A comparison of the upper and lower pairs shows that the peak distortion is greater in the a-m channels when the channels have broader bandwidths. In the f-s channels, an increase in the peak distortion occurs only when the noise signal and carrier signal are of approximately the same amplitude. In the narrow-band channels, the f-s system can tolerate noise-to-carrier ratios which are 2.5 to 4.5 db higher than for the a-m system for the same peak distortion. In the

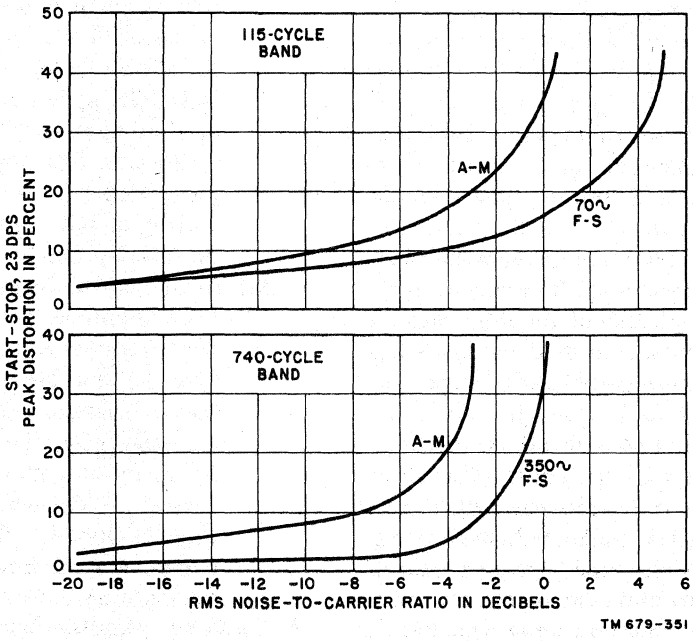


Figure 208. Distortion versus noise characteristic for a-m and f-s systems.

wide-band channels, the difference is 4 to 6 db, again in favor of the f-s system. It is this advantage of f-s systems over a-m systems that has caused a preference for f-s systems for all new carrier telegraph systems. For best performance in the presence of noise for both narrow and wide-band systems, the frequency shift should occupy 50 to 60 percent of the channel bandwidth.

174. Interchannel Interference

a. Effect of Nonlinear Elements.

(1) Interference between the channels of a multichannel system can be the result of the presence of nonlinear elements in the channels. A nonlinear element, or nonlinear operation of an element, causes signals to appear in the output of the element that are not present in the input. These output signals have frequencies that are either harmonics of the frequencies of the input signals, or sum and difference frequencies, and these frequencies can cause interference with the signals in adjacent channels. Although interchannel interference may be caused by loading coils, repeating coils, and other iron-core devices, it is usually caused by the presence of amplifiers in the circuits, particularly those amplifiers that are common to all the channels.

(2) The parasitic frequencies developed as a result of nonlinear operation generally consist of frequencies that are the sums and differences of the signal frequencies present in the common amplifiers. If the parasitic frequencies fall within the pass band of a nearby channel, distortion is introduced, and the margin against error is decreased.

(3) One method of minimizing the distortion resulting from parasitic frequencies is to restrict the amplitude of the input signal to the range over which there is linear operation of the circuit amplifiers. A second method is to use the odd harmonics of a base frequency as the channel frequencies for a-m systems. This procedure causes the sum and difference frequencies to fall halfway between the channels—that is, outside the channels, where they cannot cause interference (fig. 209). A common base frequency used in some military a-m systems is 85 cycles. The channel frequencies start with the fifth harmonic, 425 cycles. The second channel frequency is 595 cycles; that is, there is a difference of 170 cycles between the two channels. Additional channel frequencies also are separated from one another by 170 cycles up to 2,975 cycles. The four channels in figure 209 are from the carrier-telegraph system of figure 186. The channel carrier frequencies are odd harmonics of the 85 cycles base frequency.

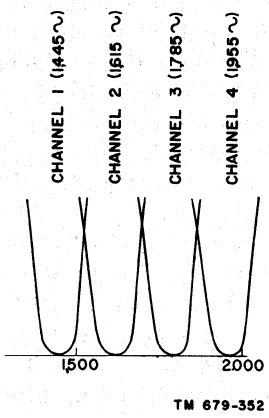


Figure 209. Placement of channel frequencies to reduce interchannel interference.

(4) In an amplifier common to all channels, a parasitic frequency may be developed which is the difference between the carrier frequencies of 425 cycles and 1,955 cycles. This frequency is 1,530 cycles. As shown in figure 209, this frequency falls halfway between the carrier frequencies of channels 1 and 2, which are 1,445 cycles and 1,615 cycles. As a result, the parasitic frequency lies outside the bandwidths of both channels, and introduces no interference in either channel.

(5) In the case of f-s systems, no means have been found as yet for using a common base frequency such as that used in a-m systems, nor has a suitable harmonic relation between carriers been found. As a result, interchannel interference tends to be more severe in f-s than in on-off systems. The effect also is augmented slightly by the fact that the average current on the line is usually greater for f-s systems than for on-off systems.

b. Spillover. Another source of interchannel interference is *spillover* into one channel by the

signals from the adjacent channels. This effect gets worse as the signaling speed increases, because of the increase in the number of sidebands, because it is the sidebands that spill over into the adjacent channels. Prevention of spillover requires effective filtering at the transmitter. It can be reduced only to a very limited extent at the receiver. Once the sidebands from adjacent channels have penetrated into a channel, the receiving filters cannot eliminate them from the channel.

175. Effects of Frequency Drift

a. Effect of Frequency Drift on A-m Systems.

(1) In any oscillator circuit, there is a tendency for the oscillator frequency to *drift* from its nominal value. The amount of drift in crystal-controlled oscillators is quite small. Only some of the more recently developed systems use crystal-controlled oscillators. The amount of drift inherent in L–C and R–C oscillators may cause an appreciable change in the oscillator frequency. The appearance of frequency drift in the oscillators in carrier telegraph channels is a major source of telegraph distortion. The drift usually is caused by the heating-up of the tube and other components of the oscillator circuit.

(2) The carrier frequency of any channel lies at the center of the channel. The filter, which limits the width of a channel, attenuates sidebands symmetrically on both sides of the carrier frequency. Should there be a change in the carrier frequency as a result of drift, the carrier frequency will no longer lie at the center of the channel, nor will the sideband attenuation be symmetrical about the displaced carrier frequency. If the carrier frequency increases as a result of drift, some of the upper sidebands, which were previously in the pass band, will lie outside the pass band, and will be filtered out. Some of the lower sidebands, which were previously outside the lower limit of the channel, will move into the pass band of the channel.

(3) The amount of bias distortion produced by frequency drift in an a-m system is relatively unimportant, however, unless the amount of drift is sufficiently great to shift the carrier frequency close to the cut-off region of the filter. The loss of an almost complete set of sidebands and the resulting decrease in the amplitude of the demodulated signal cause negative (spacing) bias to develop (D, fig. 160).

b. Effect of Frequency Drift on F-s Systems.

(1) In an f-s system, the change of carrier frequency because of drift is superimposed upon the change of frequency caused by either an M–S or S–M transition. The result is a change in the amplitude of the d-c output of the linear discriminator. The mark tuned circuit of the discriminator is resonated to a frequency above the marking frequency and the space tuned circuit is resonated to a frequency below the spacing frequency. When the oscillator frequencies drift to a higher frequency, the output from the mark circuit is greater and the output from the space circuit is less than when the oscillator frequencies were correct. This results in marking bias. When the frequencies of the oscillator decrease the result is spacing bias, because the output from the space circuit in the discriminator is greater and the output from the mark circuit is less than it was for the correct frequency.

(2) The amount of bias distortion that develops depends on the slope of the d-c pulses—that is, the time required for the transition from the spacing to the marking condition. The positive bias resulting from an increase in the amplitude of the d-c pulses is greater when there are relatively slow S–M and M–S transitions. The negative bias resulting from a decrease in the amplitude of the d-c pulses is greater when there are relatively slow S–M and M–S transitions. The frequency-drift characteristic of the TH–5/TG, which is a single-channel f-s system, is 1 percent bias for each cycle of drift of the mean carrier at a sending speed of 100 wpm.

(3) An afc (automatic frequency-control) circuit can be used to minimize the amount and the effects of frequency drift. The carrier frequencies of all the channels in a system can be controlled by comparison with a single stable frequency; or independent stable frequencies for separate

afc circuits may be included in each channel. Operating conditions and various design factors largely determine the method used.

c. Effect of Frequency Drift on Telephone Circuits.

(1) The effects of frequency drift and the need for frequency stability are particularly severe when the carrier telegraph channels are transmitted over a carrier telephone channel. The requirement of frequency stability places severe restrictions on the oscillators in both the telegraph and telephone channels. When the carrier telegraph channels are transmitted over a carrier telephone channel, the telegraph carrier frequencies and associated sidebands beat against the frequency of the channel telephone carrier. The resulting beat signals have frequencies which are the sum and difference of the telegraph and telephone frequencies. At the receiving end, the telegraph frequencies are transposed, as a group, to their original position in the frequency spectrum, by demodulation of the channel telephone frequencies.

(2) For a-m carrier telegraph systems, no appreciable distortion is introduced if the telephone oscillators at the two ends of the circuit are both kept within ± 15 cycles of the assigned carrier frequency. For f-s carrier telegraph systems, the telephone oscillators must be kept within ± 3 cycles of the assigned carrier frequencies.

176. Effect of Level Variations

a. Effect of Level Variations on A-m Systems.

(1) The remaining cause of interference and distortion in a carrier telegraph system is a *variation in the amplitude* or *level* of the carrier signal. Such variations can be the result of changes in the characteristics of the components of the systems, changes brought about by aging, or by the heating-up of the elements. The variations may cause the amplitude of the carrier signal to increase or to decrease. An increase in the amplitude of the carrier signal causes an increase in the amplitude of the d-c pulses from the detector, and results in positive bias distortion. A decrease in the carrier amplitude causes a decrease in the amplitude of the d-c pulses and negative bias distortion.

(2) In an a-m system, the operating point of the receiving relay must be near the half-amplitude point on the d-c pulses in order to obtain signals with zero bias distortion and optimum signal-to-noise conditions. A 6-db drop (reduction by half) in the level of the current causes the marking current in the receiving relay to be equal to the current in the bias winding of the relay, so that complete failure results.

(3) Agc circuits can be used to minimize the variations in the signal amplitude. When the rate at which the signal level varies is relatively slow compared with the signaling speed, the agc circuit can be used to maintain a nearly constant level into the demodulator. The time constant of the R–C circuit in the agc system must have a value that enables it to follow the level variations. When the signal level varies slowly, the time constant can be long compared with the duration of a signaling element. When the signal level varies rapidly, the time constant must be short to be able to follow the signal-level variations. However, the use of a short time-constant R–C circuit in the agc system leaves the receiving amplifiers open to noise in the spacing interval. A-m carrier telegraph systems are impractical where large, rapid variations in the signal level occur.

b. Effect of Level Variations on F-s Systems. No bias distortion is produced in an f-s system by sudden changes in the signal level, as long as the levels remain within the range over which the limiter presents a constant amplitude signal to the discriminator.

177. Speech-Plus-Duplex Systems

a. Purpose of Speech-Plus-Duplex Systems.

(1) Simultaneous speech and teletypewriter communication is achieved by the use of *speech-plus-duplex systems*. This method of operation is used when it is desirable to set up a telegraph circuit quickly, because it makes use of an already existing telephone circuit. It is used only when it is difficult to set up the

telegraph circuit by any other means, because of the slight impairment of the speech signal produced by the removal, from the telephone signal, of the portion of the v-f range required for the telegraph signals.

(2) A portion of the 200- to 3,000-cycle v-f range is used for a v-f teletypewriter channel. A, figure 210, illustrates one manner in which the frequency range is divided. The band between 200 and 2,000 cycles is used for telephone transmission. The band between 2,000 and 3,000 cycles is used for carrier telegraph transmission. The attenuation characteristic of the telephone filter is shown in B. The filter passes the frequencies in the telephone pass band, and rejects those in the telegraph pass band. In C, it is shown that the telegraph filter passes the frequencies in the telegraph pass band and rejects those in the telephone pass band.

(3) In figure 211, A demonstrates the second manner in which the v-f range is divided for speech-plus-duplex operation. A slot in the v-f range is used for carrier telegraph transmission. The bandwidth of the slot must be such that its removal from the v-f range causes least impairment of the voice signal. For this reason, the band from 1,500 to 2,000 cycles is used. The telephone filter, shown in B, removes the 500-cycle slot from the telephone-frequency spectrum. The telegraph filter, in C, limits the telegraph channel to the 1,500- to 2,000-cycle range. In this slot, transmission in one direction is at a frequency of 1,680 cycles and in the opposite direction it is at 1,860 cycles. As usual, the carrier signal is present for marking pulses, and no signal is present during the spacing terminals. The range between 1,150 and 1,350 cycles also is used frequently as the slot within which carrier telegraph transmission takes place. This slot is used when it is desired not to interfere with the ringing frequencies, which may be 1,600 or 2,000 cycles.

b. Description of Speech-Plus-Duplex System. The components of a speech-plus-duplex system are illustrated in figure 212. The telephone circuit makes use of a ringing circuit operating at frequencies which do not interfere with the telegraph channel. The telephone filter removes from the v-f range the slot to be used for telegraph transmission. For sending telegraph messages, the oscillator-modulator is controlled by a polar relay. The sending filter limits the telegraph channels to a band of frequencies about the 1,680-

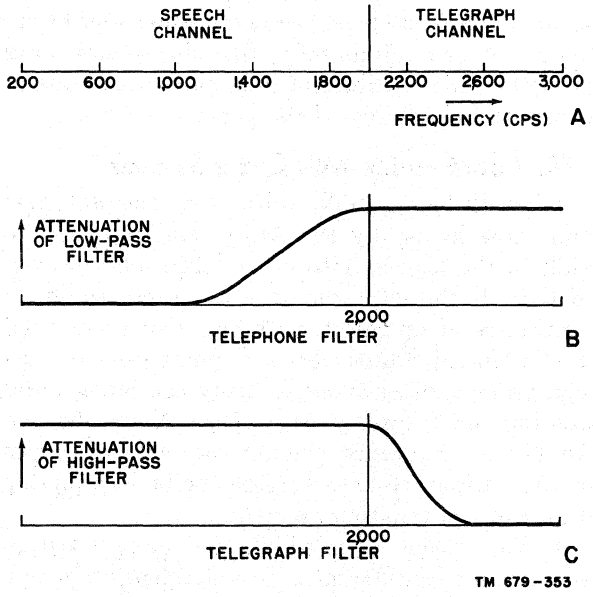


Figure 210. Frequency allocation in speech-plus-duplex systems.

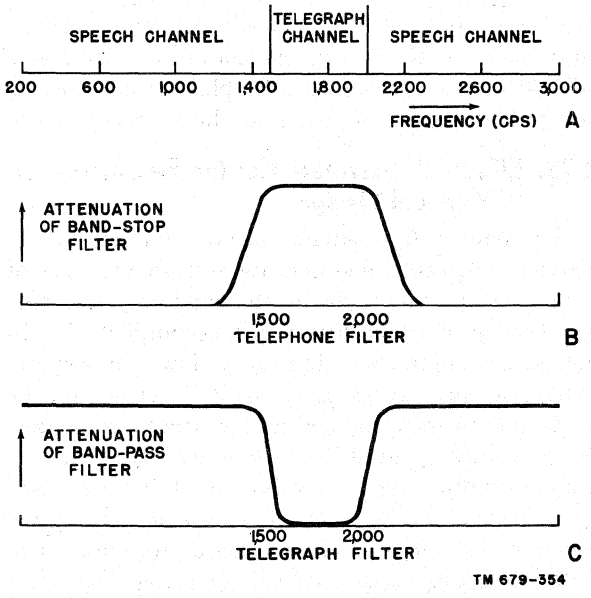


Figure 211. Alternative frequency allocation in speech-plus-duplex system.

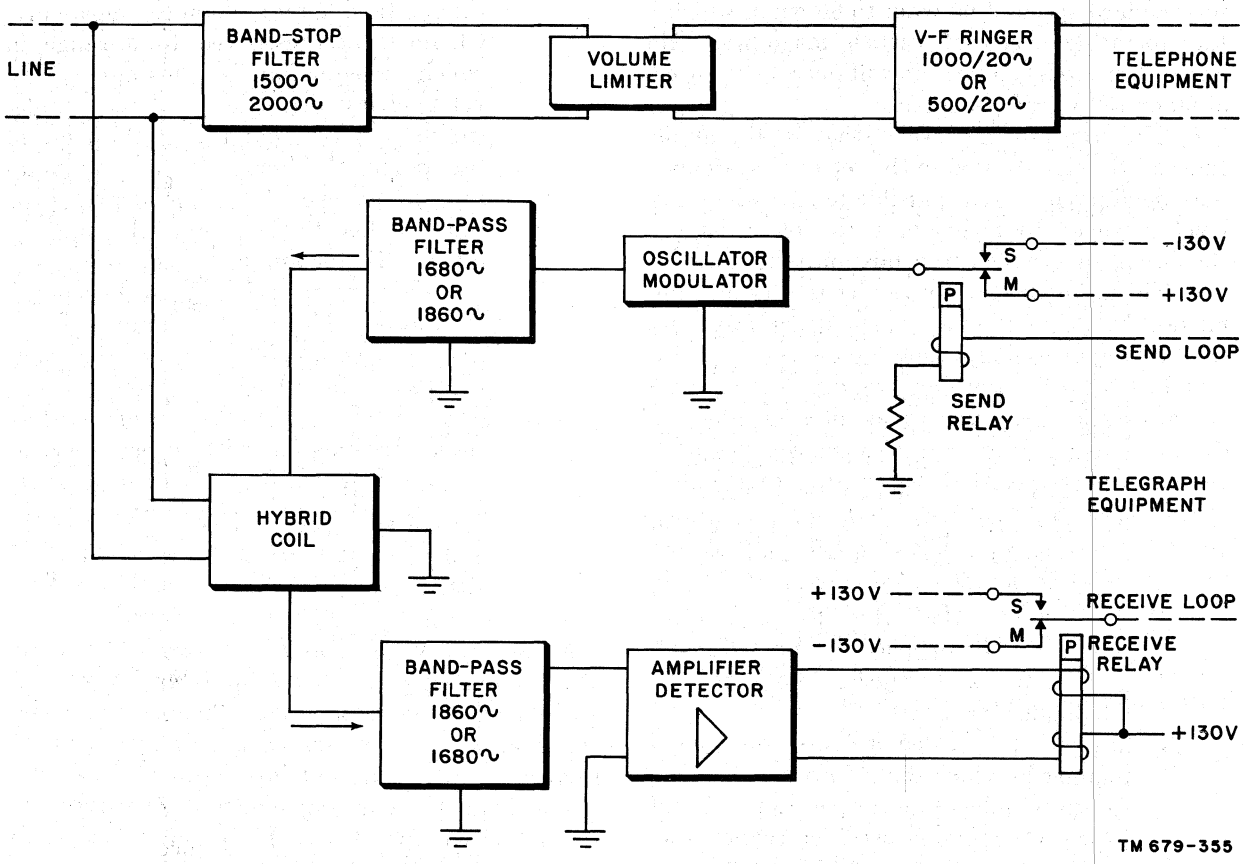


Figure 212. Speech-plus-duplex system.

cycle and 1,860-cycle carrier frequencies. The receiving filter selects, from all the input signals, only those in the telegraph channels. After amplification and detection take place, the resulting d-c pulses are transmitted to the receiving loop.

178. Effect of Selected Carrier Frequency on Physical Design

The carrier frequencies selected for use in the carrier telegraph channels determine the size of many of the elements in the channel, and also the ease with which filtering is accomplished. In telegraph channels using very low carrier frequencies, large-sized parts must be used, for the coils and capacitors used at low frequencies have large values of inductance and capacitance. At higher frequencies, the values of inductance and capacitance can be reduced, so that smaller sized parts can be used. If the carrier frequencies in different channels are spaced far apart, there is a reduction in the number of sideband frequencies which might interfere with the adjacent channels. As a result, the limitations on the width of a channel need not be as severe as with narrow band channels, and the necessary amount of filtering is reduced. Indirectly, this also makes for a reduction in the values of inductance and capacitance, and in the size of the parts.

179. Coordination with Other Systems

It would be desirable to link the telegraph systems now in use by the Army, and those to be built in the future, with other telegraph systems; so that, if the need should arise, there could be continuity of operation with the civil plant used in the United States, the civil plant used by foreign nations, older types of Army equipment, and existing and future systems of the Navy and the Air Force. However, the different characteristics of the various systems impose limitations on the degree of coordination possible.

a. The speeds with which the teletypewriters operate may be different. British teleprinters and American teletypewriters may be operated on the same circuit but, to make this possible, it is necessary to speed up the American teletypewriter from

its standard speed of 60 wpm to 66 wpm, which is the operating speed of British machines. The newer Army teletypewriters will operate at speeds of 60 or 100 words per minute.

b. The transmitting levels—that is, the amplitudes of the signals used in the separate systems—may be different. It is possible to reset the signal level, however, by means of a variable resistor in the circuit over which transmission takes place.

c. The methods used for carrier telegraph transmission may differ, one system using frequency shift and another amplitude modulation. A link may be established between two different methods of transmission, however, by use of a repeater. If the transmission is to be from a frequency-shift to an amplitude-modulated system, the input circuit of the repeater would contain a receiver for frequency-shift signals. The resulting d-c pulses would control the operation of a modulator in an amplitude-modulated carrier transmitter.

d. The carrier frequencies in the telegraph channels of the systems to be linked may be different. This would require changes in the oscillators producing the carrier frequencies in one of the systems. It also would require changes in the filters of this system.

e. The widths of the channels in the systems to be linked may be different. Coordination would be achieved, in this case, by changing the pass bands of the filters in one of the two systems.

180. Summary of Characteristics of Carrier Telegraph Systems

a. Transmission Objectives.

(1) It should be possible to maintain serviceable communication at all times.

(2) The size and weight of the equipment should be limited.

(3) The speed of spending should not be too slow. The build-up time of a pulse is determined by the bandwidth of the channels and filters through which the signal passes. If the buildup time of the pulses is long, fast sending is prevented, because characteristic distortion would result. For fast sending, broad bandwidth channels and filters must be used. Narrow-band channels and filters limit the speed of sending by increasing the buildup time.

(4) There should be a maximum number of channels. The frequency-attenuation characteristic of the range filter may attenuate different channels unequally. Channels are restricted to a range in which attenuation is constant. Envelope-delay distortion makes it undesirable to have channels at the edge of the range of filters. If the minimum carrier frequency is too low, it places the channel close to the edge of the range of filters. Distortion may also result because of fortuitous changes in the lengths of pulses. If the maximum carrier frequency is too high, it places the channel close to the edge of the filter range.

(5) There should be a minimum of interference. Interaction with other services arises because the frequencies of the services are too close, and the power levels interfere. This type of interference is minimized by coordination of the services. Noise signals produce changes in the amplitude and frequency of carrier signals. The amplitude changes produce interference in a-m systems. The frequency changes produce interference in f-s systems. The effects of noise are greater on a-m than in f-s systems. Interchannel interference is produced by nonlinear operation of the elements in a telegraph system. Parasitic frequencies are developed which lap over into adjacent channels. Interference of this type is minimized by linear operation of amplifiers and by causing all carrier frequencies to be odd harmonics of a base frequency. Frequency drift is a change in oscillator frequency. It causes negligible distortion in a-m systems, until the carrier reaches the cutoff region of the filter. It changes the amplitudes of the d-c pulses from the discriminator in f-s systems, and produces distortion. Level variations are produced by changes in the characteristics of circuit elements. Changes in the amplitudes of the d-c pulses from the detector in a-m systems are produced by level variations, and distortion results. Such variations are minimized by automatic gain-control circuits. The effects of level variations are negligible in f-s systems, for the limiter minimizes the level variations.

b. Speech-plus-Duplex Systems. This method of operation permits simultaneous speech and tele-

typewriter communication. In the 200- to 3,000-cycle voice-frequency range, the bandwidth from 200 to 2,000 cycles is used for telephone transmission, the remainder for telegraph transmission; or the band between 1,500 and 2,000 cycles is eliminated from the telephone range of frequencies and devoted to telegraph transmission.

c. Coordination with Other Systems.

(1) It should be possible to link present and future Army teletypewriters with the civil plant used in the United States; the civil plant used by foreign nations; older types of Army equipment; existing and future systems of the Navy and the Air Force.

(2) Limitations on the degree of coordination are caused by differences in sending speeds, signal levels, methods of transmission, carrier frequencies, and channel bandwidth.

181. Review Questions

a. Define: telegraph; teletypewriter.

b. Define: direct-current telegraphy; carrier telegraphy; radio telegraphy; marking signal; spacing interval.

c. What functions are performed by the transmitter in a teletypewriter? What functions are performed by the receiver in a teletypewriter?

d. What are the differences between full-metallic and ground-return teletypewriter circuits? What are the advantages of both types of circuit? The disadvantages?

e. What are the advantages of teletypewriter operation?

f. Define: neutral operation; polar operation; half-duplex operation; full-duplex operation: How is neutral operation accomplished? Polar operation?

g. Define: loop operations; trunk operations. Why are repeaters used in telegraph transmission circuits? What is meant by neutral to negative battery? Neutral to positive battery? Why is proper connection of relays in such circuits important?

h. How is two-path polar operation accomplished? How does differential sending differ from polar sending? What are the advantages of polarential operation over half-duplex neutral operation?

i. What are the characteristics of a theoretical waveshape in a neutral teletypewriter system? In a polar system?

j. What are the characteristics of an actual waveshape in a neutral teletypewriter system? In a polar system?

k. Define: bias distortion; positive bias distortion; negative bias distortion.

l. Define: the operating current of a relay; the release current. Describe the effects of the inductance and distributed capacitance in the circuits on the delay of the operating and release points of a relay. Describe the effects of a change in the line current; effects of leakage along the line.

m. What are the properties of characteristic distortion? Define positive characteristic distortion, negative characteristic distortion. Describe how negative characteristic distortion is produced. Describe how positive characteristic distortion is produced.

n. Define fortuitous distortion. What are th causes of fortuitous distortion?

o. Define the selecting interval. What is its position in a perfect signal pulse? What is the margin against error? Describe how the displacement of a signal transition resulting from bias distortion can cause a printing error.

p. How can the bias relay minimize bias distortion? State the purpose of a rangefinder. Briefly describe how it is used.

q. What are the advantages of carrier telegraphy? What are the different methods used for carrier-telegraph transmission? What are the differences in the frequency bandwidths used by the various methods?

r. What are the essential components of an amplitude-modulated on-off system? Draw a block diagram showing how the components are connected. State briefly the purpose and characteristics of each component. Describe briefly how each component operates.

s. What are the essential components of a frequency-shift system? Draw a block diagram showing how the components are connected. State briefly the purpose and characteristics of each component. Describe briefly how each component operates.

t. What are the transmission objective in a carrier-telegraph system? State the relationship between the buildup time of a pulse and the bandwidth of the filters through which the pulse passes. State the relationship between the buildup time of a pulse and the sending speed of a teletypewriter. State the relationship between the sending speed

of a teletypewriter and the pass band of the filters in a teletypewriter channel.

u. In what way is the number of available telegraph channels limited by envelope-delay distortion? The minimum carrier frequency? The maximum carrier frequency?

v. How is interference caused in a telegraph system by interaction with other services? Noise signals? Interchannel interference? Frequency drift? Level variations? In what manner does each of these sources of interference affect a-m systems? F-s systems? How is the effect of each of these sources of interference minimized?

w. What is the purpose of a speech-plus-duplex system? When is such a system used? How are the telephone and telegraph channels separated? Draw a block diagram of such a system.

x. Describe the effects of the selected carrier frequencies on the size of the parts in a carrier-telegraph system; on the filtering.

y. With what other systems should it be possible to coordinate Army teletypewriters? What are the limitations on such coordination? Describe briefly how each limitation can be overcome.

CHAPTER 6
TRANSMISSION MEASUREMENTS

Section I. TRANSMISSION UNITS

182. Need for Transmission-Performance Measurements

A transmission line may cause deterioration of a transmitted signal in many ways—by frequency versus attenuation characteristics of a transmission line; by attenuation of its power; by distortion of its waveshape; and by permitting interference. Methods must be devised to minimize these types of deteriorations.

a. Suppose a repeater is to be installed somewhere on the line to make up for the power loss. The amount of amplification to be provided by the repeater should be determined by the amount of power loss. It would be inefficient to keep rebuilding a repeater until it provided the necessary amount of amplification; but, if the amount of power loss could be ascertained, the required amount of amplification would be known, and it would become easy to build the repeater to provide the amount of amplification necessary to make up the loss. To do this, there must be a way to measure and express *numerically* the power loss, distortion, crosstalk of a transmission line, the power gain of an amplifier, and the performance characteristics of components such as filter and equalizer networks that might be used with them.

b. In this chapter, a system of measurement is developed and applied to transmission lines. Here the input and output powers of a circuit can be measured, and their ratio expressed in terms of a unit called a *decibel.*

183. Units of Transmission Measurements

a. Bel. The ratio of the input and output powers was formerly expressed in terms of a unit called the *bel,* a name derived from Alexander Graham Bell, the inventor of the telephone. With this unit, the output power level is said to be 1 bel below the input level when the input and output powers are in the ratio of 10 to 1. In this case, the bel is used as a unit of *attenuation.* If the output power level is 10 times the input power level, it is said to be 1 bel above the input. In this case, the power ratio of input to output is 1 to 10, and the bel unit is a *gain.*

b Decibel.

(1) The bel proved to be an inconveniently large unit. It was found desirable to use a unit which more closely expresses commonly encountered (smaller) power ratios without using fractional parts of the unit. For this reason, the *decibel* has been universally adopted. As its name implies, a decibel (*db*) is one-tenth of a bel. Today, power ratio is expressed as a definite number of decibels. Because the decibel is a unit one-tenth as large as the bel, the number expressing a power level in decibels (abbreviated N_{db}) is 10 times as large as the number of bels. The formula for power level in decibels is—

$$\text{number of decibels, } N_{db} = 10 \log_{10} \frac{P_1}{P_2}.$$

(For reference to logarithms, see app.) Applying this formula, a power ratio of 100 to 1 is expressed as 20 decibels, or 20 *db,* instead of 2 bels; a power ratio of 2 to 1 is expressed as 3.01 decibels, instead of 0.3010 bel.

(2) It should be noted that although the decibel is one-tenth of a bel, the power ratio expressed by 1 decibel is *not* one-tenth of the power ratio expressed by 1 bel. One decibel expresses a power ratio of 1.26, because the logarithm of 1.26 is .1, and the number of decibels is 10 times this logarithm.

c. Meaning of db. The decibel is a unit of power *ratio,* not of power. When a transmission line causes 10 db of attenuation, it means that the

output power is only one-tenth of the input power. It does *not* state the *amount* by which the power has changed. If the input power is 100 watts and the output power is 10 watts, the line has absorbed 90 watts of power. If the input power is 1 watt and the output power is 0.1 watt, the line has absorbed only 0.9 watt of power. In both cases, the power ratio is 10 to 1, and the attenuation is 10 db.

184. Use of Decibel

a. Conversion of Power Ratio to Decibels.

(1) The decibel, like the bel, may be used as a unit either of attenuation or gain, expressing the decrease in power caused by a transmission line or by a piece of equipment, or the increase in power caused by an amplifier. In order to distinguish whether a given number of decibels of power change refers to gain or loss, the number of decibels customarily is preceded by a plus sign or a minus sign. Thus, a power change of $+6$ *db* refers to a gain, and -6 *db* refers to a loss.

(2) In converting a power ratio into its decibel expression, the following two rules should be followed. *Rule 1:* In the formula,

$$N_{db}=10 \log \frac{P_1}{P_2},$$

choose the larger power as P^1 and the smaller power as P_2, regardless of whether the larger power is at the input or the output. The power ratio then will always be greater than 1, and its logarithm will be a positive number. *Rule 2:* Prefix to the answer a plus sign if the power change is a gain, or a minus sign if the power change is a loss.

(3) For example, the power input to a transmission line is 10 milliwatts, and the power output is 2.46 milliwatts. By rule 1, the larger power (the input) is P_1, and the smaller power is P_2. The number of decibels of power change then is expressed as—

$$N_{db}=10 \log \frac{P_1}{P_2}=10 \log \frac{10.00}{2.46}=10 \log 4.06$$

(Note that the subscript 10, denoting the base of the common system of logarithms, is omitted. This is general practice when no misinterpretation is possible, and is followed throughout.) Reference to a table of common logarithms shows that the logarithm of 4.06 is 0.6091. Hence,

$$N_{db}=10 \log 4.06=10 \times .6091=6.091$$

Since the input power is greater than the output power, the power change is clearly one of attenuation. By rule 2, the answer is prefixed by a minus sign, thus:

$$N_{db}=-6.091 \ db$$

(4) As an example of a power gain, consider a repeater amplifier having an input power of 2 milliwatts and an output power of 400 watts. By rule 1, the larger power (the output) is P_1, and the smaller power is P_2. Both powers must be expressed in the same units. Hence, the input power of 2 milliwatts is expressed as 0.002 watt. The power change is then calculated:

$$N_{db}=+10 \log \frac{P_1}{P_2}=+10 \log \frac{400}{0.002}$$

$$=+10 \log 200{,}000$$

$$=+10 \times 5.3010=+53.01 \ db$$

In this example, as a short cut, the plus sign (indicating that the power change is a gain) is inserted at the start, and carried through to the final answer.

(5) It should be noted that whether a power change is a loss or a gain, it is necessary to know the *power* ratio. In the case of an amplifier, its *power gain* is the power ratio required. The voltage gain of an amplifier is *not* the same as the power gain.

b. Conversion of Decibels to Power Ratio.

(1) If it is required to find the power ratio of a circuit for which the gain or loss in decibels is known, a reversal of the procedure described in *a* above is used. The following rule applies. *Rule 3:* If the number of decibels is *positive*, the circuit

has a power *gain*, and the output power is greater than the input power; if the number of decibels is *negative*, the circuit has a power *loss*, and the output power is less than the input power.

(2) As one example of how to determine the ratio of output and input powers when the ratio is expressed in decibels, assume there is a circuit known to have a power change of +12 *db*. Inserting this value in the formula, and dividing both sides by 10:

$$12\ db = 10 \log \frac{P_1}{P_2}$$

$$1.2 = \log \frac{P_1}{P_2}$$

A table of logarithms shows that 15.85 is the number which has the logarithm 1.2. Therefore,

$$\frac{P_1}{P_2} = 15.85$$

Since the number of decibels is given as positive, it is known from rule 3 that the circuit has a gain, and that its output power is 15.85 times its input power.

(3) As a second example, consider a power change of −25 *db*. The minus sign is disregarded temporarily, and the power ratios found in the manner explained in (2) above:

$$25\ db = 10 \log \frac{P_1}{P_2}$$

$$2.5 = \log \frac{P_1}{P_2}$$

$$\frac{P_1}{P_2} = 316.2$$

Since the number of decibels has a minus sign, it is known from rule 3 that the circuit attenuates power, and that its output power is less than its input power by a ratio of 1 to 316.2.

c. Addition of Power Changes.

(1) The total number of decibels of attenuation or gain of a circuit is the algebraic sum of the attenuations or gains of its stages or sections, each expressed in decibels. For example, consider the transmission-line network of figure 213. There is a power loss of −10 *db* in the first section of the line, a power loss of −20 *db* in the equipment between the sections of the line, a power loss of −10 *db* in the second section of line, and a power gain of +30 *db* in the repeater. The total power change is—

$$N_{db} = -10\ db - 20\ db - 10\ db + 30\ db = -10\ db$$

The total power change is −10 *db*, indicating a net power loss in a ratio of 10 at the input to 1 at the ouptut

(2) Reference to table IV provides an explanation of this result. Each −10-*db* loss represents a power-loss ratio of 10 to 1. The −20-*db* loss represents a power loss ratio of 100 to 1. The +30-*db* gain represents a power gain ratio of 1 to 1,000. The overall loss is the product of these ratios:

$$\text{Overall loss} = \frac{10}{1} \times \frac{100}{1} \times \frac{10}{1} \times \frac{1}{1000} = \frac{10}{1}$$

An overall loss in a ratio of 10 to 1 is expressed as −10 *db*, the same result obtained by algebraic addition of the decibel losses and gains of the circuit sections.

185. Table of Decibels

a. In order to eliminate the need for extensive calculations and reference to a table of logarithms each time a power ratio must be expressed in terms of decibels, reference may be made to a

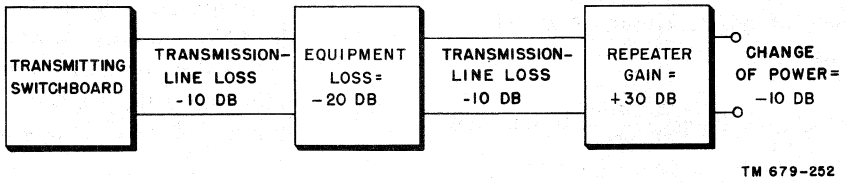


Figure 213. Power changes in transmission-line network.

table, such as table IV, giving power ratios *directly* from decibels, and vice versa. Table III is such a table.

Table IV.

Number of decibels	Losses		Gains	
	Power-ratio factor	% power loss	Power-ratio factor	% power gain
1	0.79	21	1.26	26
2	.63	37	1.6	60
3	.5	50	2.0	100
4	.4	60	2.5	150
5	.32	68	3.2	220
6	.25	75	4.0	300
7	.2	80	5.0	400
8	.16	84	6.0	500
9	.126	87.4	7.9	690
10	.1	90	10.0	900
20	.01	99	100.0	9,900
30	.001	99.9	1,000.0	99,900
40	.0001	99.99	10,000.0	999,900

b. The manner in which table IV can be used is illustrated by a consideration of all the figures on one line.

(1) Consider the case for which $N_{db}=3$.

$$3=10 \log \frac{P_2}{P_1}$$

$$.3=\log \frac{P_2}{P_1}$$

.3 is the logarithm of the number 2; consequently,

$$\frac{P_2}{P_1}=2$$

The *power-ratio factor* (table IV) is the ratio of the output and input power. In the example above, if there is a power *gain*, the power-ratio factor is 2; the output power P_2 is greater than the input power P_1 in a 2-to-1 ratio. If there is a power *loss*, the power-ratio factor is 0.5, because the output power must be less than the input power in a 1-to-2 ratio. The table quickly gives the power-ratio factor for both gains and losses.

(2) Both the *percentage-power loss* and the *percentage-power gain* are the *ratio* of the *difference* between the input and output powers, and the *input power*, expressed as a percentage. Example of percentage-power loss: The input power to a transmission line is 10 milliwatts, and the output at a receiving switchboard is 5 milliwatts. The percentage-power loss is

$$\frac{\text{input power–output power}}{\text{input power}} \times 100$$

Substituting,

$$\frac{\text{10 milliwatts–5 milliwatts}}{\text{10 milliwatts}} \times 100$$

$$=50 \text{ percent}$$

The percentage-power loss is 50 percent. (The power-ratio factor is 0.5, and there is a 3-*db* power loss.) Example of percentage-power gain: The input power to a transmission line is 7 milliwatts, and the output is 14 milliwatts. The percentage-power gain is

$$\frac{\text{output power–input power}}{\text{input power}} \times 100$$

Substituting,

$$\frac{\text{14 milliwatts–7 milliwatts}}{\text{7 milliwatts}} \times 100$$

$$=100 \text{ percent}$$

The percentage-power gain is 100 percent. (The power-ratio factor is 2, and there is a 3-*db* power gain.)

c. Table IV can be used also for figures that do not appear in it. For example, consider a power change of 24 *db*. A power change of 24 *db* is equal to the sum of 20 *db* and 4 *db*. According to the table, 20 *db* corresponds to a power change in the ratio 100 to 1, and 4 *db* corresponds to a power change in the ratio 2.5 to 1. As a result, 24 *db* corresponds to 100×2.5 to 1, or 250 to 1. For gains, the power-ratio factor is 250 to 1; for losses, the power-ratio factor is 0.01×0.4 to 1, or 0.004 to 1, or 1 to 250. For gains, the percentage-power gain is (250 − 1)×100=24,900 percent; for losses, the percentage-power loss is (1 − 0.004)×100= 99.6 percent.

186. Other Uses of Decibels

Heretofore, the decibel has been considered only as a measure of power losses caused by attenuation and power gains resulting from amplifi-

cation. But it is used in the measurement of three other types of power loss that can occur as the result of the insertion of equipment into, or across, the transmission line—insertion loss, mismatch or reflection loss, and bridging loss.

a. Insertion Loss. Insertion loss is the ratio of the powers measured at the receiving end of a transmission line after and before a network has been inserted somewhere in the system. In A, figure 214, the power transmitted directly to the receiving end is P_1. In B, a network has been inserted into the line; the network causes a change of power, so that the power at the receiving switchboard after and before its insertion is P_2/P_1, and the insertion loss expressed in decibels is

$$N_{db} = -10 \log \frac{P_2}{P_1}$$

An insertion loss can be caused by the presence of a filter in the line. When the power ratio is expressed as a positive number of decibels, an insertion gain, rather than an insertion loss, is indicated.

b. Mismatch or Reflection Loss. *Mismatch loss* or *reflection loss* is the ratio of the power measured when there is a mismatch at the receiving end of the transmission line (or elsewhere on the line), to the power measured at the receiving end when the receiver is matched to the line (and there is no mismatch elsewhere on the line). In A, figure 215, the receiver is matched to the line, and consequently the impedance at every point on the line is equal to the characteristic impedance of the line, and there is maximum transfer of power to the receiving switchboard. This amount of power is P_1. In B, a cable of different characteristic impedance has been inserted into the line, so that a mismatch exists at both junctions of the cable and the line. As a result of the mismatch, a portion of the transmitted power is reflected, reducing the power at the receiving switchboard. This reduced power is P_2. The ratio of the powers measured at the receiving end without and with matching on the line is P_2/P_1, and the reflection or mismatch loss expressed in decibels is

$$N_{db} = -10 \log \frac{P_2}{P_1}$$

A reflection loss can be caused by a mismatch occurring at the junction of the transmission line and a cable or a piece of equipment inserted *into* the line, or to a change in the properties *of* the line, or to a mismatch *between* the receiver and the line.

c. Bridging Loss. Bridging loss is the ratio of the powers measured at the receiving end of a transmission line after and before a device has been connected or bridged *across* the line. In A, figure 216, the power transmitted directly to the receiving end is P_1. In B, a transmission measuring set, connected across the line, has drawn some of the power from the line, so that the power at the receiving end now is P_2. The ratio of the powers at the receiving switchboard after and before the measuring set was put into the line is P_2/P_1, and the bridging loss expressed in decibels is

$$N_{db} = -10 \log \frac{P_2}{P_1}$$

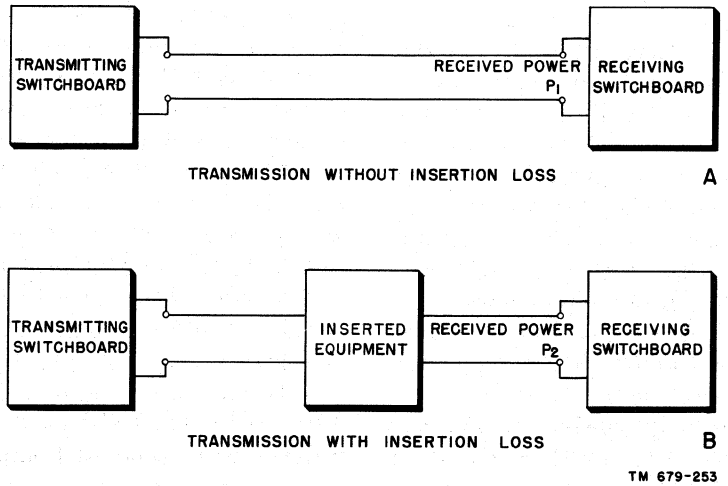


Figure 214. Insertion loss.

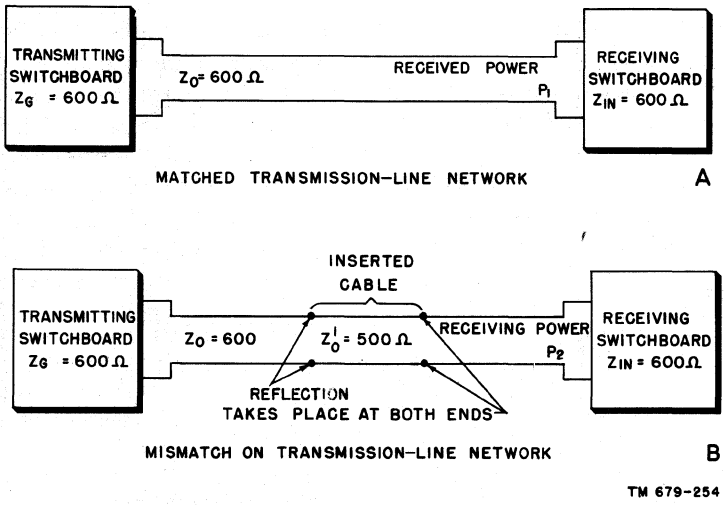


Figure 215. Mismatch or reflection loss.

It is usually desirable that the bridging loss resulting from the connection of a measuring set across a transmission line be kept to a minimum. This means that the measuring set must draw minimum power from the line.

187. Use of Standard Testing Power

a. The decibel is fundamentally a unit of power ratio, rather than power (par. 183 *c*). This results in a disadvantage in its use. For example, 10 *db* indicates that there has been a change of power by a factor of 10, but there is no indication of what the amount of change is. The values of the initial and final powers are not given by the decibel as a unit of attenuation or gain. It would be advantageous to be able to use decibels not only as the measure of a power ratio but also as the measure of a specific amount of power. This is done in the following example. In the equation, where

$$N_{db}=10 \log \frac{P_2}{P_1}$$

Then

$$\text{if } P_1=1,$$

$$N_{db}=10 \log P_2.$$

Consequently, if the input power has a value of unity, and if P_2 is the output power, N_{db} is 10 times the logarithm of the output power. When the number of decibels is given. the output power can be ascertained quickly, as the number of decibels refers to the ratio of the output power to 1. On a transmission line, the power transmitted is very small, being measured in milliwatts. Therefore, the reference or input power level has been chosen arbitrarily as 1 milliwatt. Then, $N_{db}=10$ indicates that there has been a gain of 10, and the output power is 10 milliwatts; $N_{db}=20$ indicates that there has been a gain of 100, and output power is 100 milliwatts. But there must be some way of indicating the difference between N_{db} used as the expression of a power ratio and N_{db} used as an expression of an absolute amount of power. To permit indicating this difference, *db* is used only as the expression of a power ratio, and *dbm* is used as the measure of absolute power as compared to an arbitrary reference level of 1 milliwatt; 1 milliwatt of power is equal to 0 *dbm*. $N_{db}=10$ therefore represents a change of power by a factor 10, and $N_{dbm}=10$ represents 10 milliwatts of power.

b. In telephone transmission, it has been found that the elimination of all frequencies below 200 cycles and above 2,700 cycles per second entails very little loss in the intelligibility of the received signal. Most voice-frequency equipment is designed to transmit effectively this 200- to 2,700-cycle band, known as the voice-frequency range. A 1,000-cycle sine-wave generator commonly is used as a standard for checking and calibrating all types of equipment. This calibrating frequency is near the middle of the voice-frequency band. In order to measure power loss or gain in a speech circuit, a 1,000-cycle sine-wave tone of 1-milliwatt power is used as a standard testing signal.

c. It must be emphasized that the measurement of power in *dbm* is independent of a standard impedance or a standard frequency. Although the testing signal for voice-frequency circuits is a 1,000-cycle tone, the *dbm* can be used for measur-

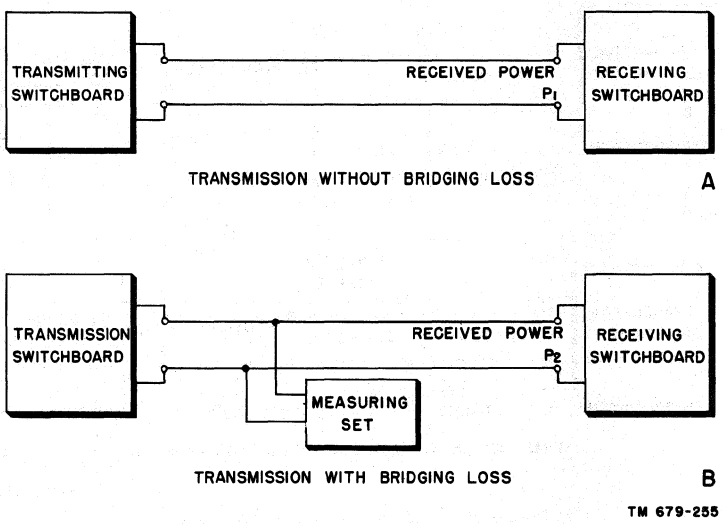


Figure 216. Bridging loss.

ing power at any frequency in any type of circuit, so long as the reference power is understood to be 1 milliwatt (0 *dbm*).

188. Zero-Level Point

a. The telephone circuit in A, figure 217, consists of two switchboards connected by a transmission line. A repeater is used in the line to make up for the power loss caused by attenuation. When a 1,000-cycle test tone of 1-milliwatt power is used to measure the power loss or gain in a speech circuit, the point at which the test-tone signal is applied is the reference point. The transmission switchboard shown in A is the reference point; consequently, there is 0-*dbm* absolute power. This point is referred to as the 0-level point.

b. The west section of the transmission line, shown in A, causes 10 *db* of power loss, the repeater causes 30 *db* of power gain, and the east section of the transmission line causes 10 *db* of power loss. These power changes are expressed in *db* (not *dbm*) when they are with respect to themselves—that is, along the line between switchboards; the test tone powers are expressed in *dbm* (not *db*) when they are with respect to the fixed 1-milliwatt level. The total power change in *db* (the algebraic sum of all the power changes in *db*) is +10 *db*. This total power change is with respect to the 1-milliwatt level at the transmitting switchboard, so that at the input of the receiving switchboard the test tone power is +10 *dbm* above the zero-level point at the transmitting switchboard.

c. In summarizing this, the following points should be remembered: Zero *dbm* indicates a power level of 1 milliwatt. A positive number of *dbm* indicates a power level of greater than 1 milliwatt. A negative number of *dbm* indicates a power level less than 1 milliwatt. It is desirable that in all communication systems the zero-level point be the same; and it is taken to be the transmitting switchboard.

189. Power-Level Diagrams

The mathematical results in paragraph 188 can be represented as a graph called a power-level diagram, also called an energy-level diagram. Such a diagram shows the absolute powers throughout a circuit graphically, and is used in both the design and maintenance of long lines using repeaters.

a. B, figure 217, represents the power-level diagram for the one-way telephone system shown in A. The horizontal base line indicates a level of 0 *dbm*. Each horizontal line above and below represents a value of 1 *dbm*. Positive values are located above the base line, negative values below it. At the transmitting switchboard there is a 0-*dbm* level. Over the distance between the transmitting switchboard and the repeater, the power decreases 10 *db* below 0-*dbm* level. The repeater increases the power 30 *db*. This increase is represented by a vertical line because it does not take place over a distance, as does the power loss between switchboard and repeater. Between the repeater and the receiving switchboard, there is again a decrease of 10 *db* in the power level. A line representing this loss is drawn downward 10

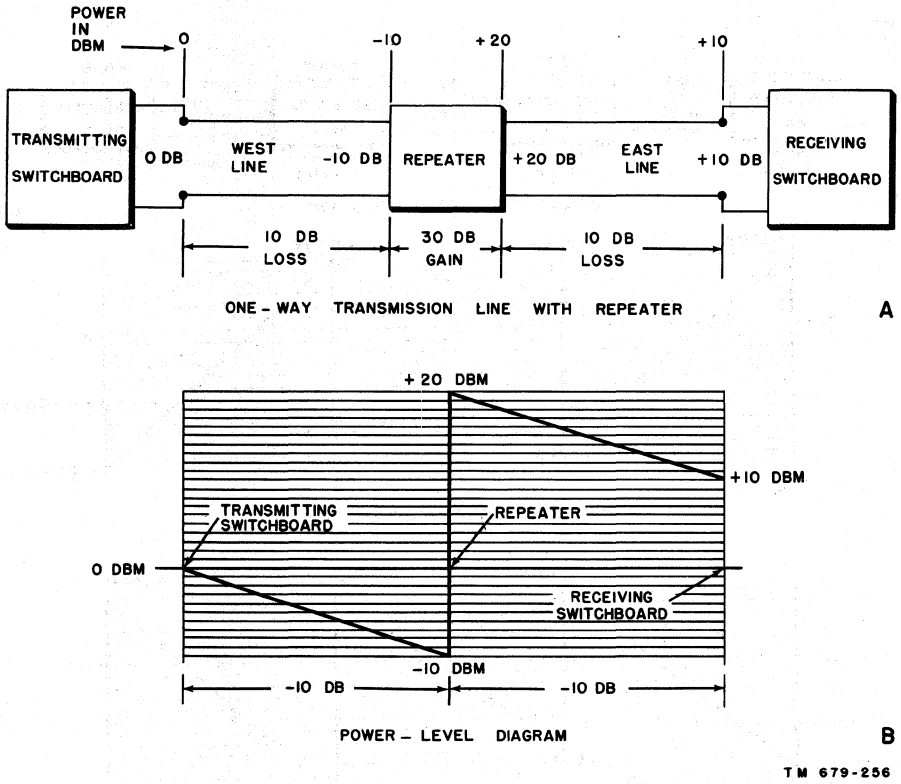


Figure 217. One-way transmission line with power-level diagram.

horizontal lines from the +20 *db* level at the output of the repeater to the +10 *db* level at the input of the switchboard. The final result indicates that between the switchboards the test tone power has increased to 10 *dbm*, so that the test tone input to the receiving switchboard is 10 milliwatts.

b. A two-way transmission line is shown in A, figure 218. The four repeaters in the line pass the signal in both directions, as indicated by the arrows. The values above and below the arrows show the gains of the repeaters in both directions.

(1) B shows the power levels in transmission from west to east. At transmitting switchboard A, the power is at 0 level (1 milliwatt). Power gains caused by the four repeaters are represented by vertical lines proportional in length to the gains. Thus, at repeater No. 2 the input power is −10 *dbm* (10 *db* down from the 0-level point). The line indicating the lower gain caused by the repeater goes from −10 *dbm* up through 16 horizontal lines to a power of +6 *dbm*, because the gain of the repeater is +16 *db*. Power losses resulting from attenuation along the transmission lines are represented by lines drawn diagonally downward. The vertical component of the lines indicates a lowering of the power; the horizontal component indicates that the power losses take place over the distances between repeaters. Thus, the power loss between repeaters Nos. 2 and 3 is represented by a line extending from the +6 *dbm* level at the output of the second repeater downward through 14 horizontal lines to a power of −8 *dbm*, because the transmission line has caused a power loss of 14 *db*. At the input to repeater No. 3 the power is −8 *dbm*.

(2) Figure 218, view C, shows the power levels in transmission from east to west. The diagram is drawn in the same manner as in view B; since switchboard B is now the transmitting end, it is the 0-level point, and the diagram is read from right to left.

c. B and C use a test tone input power of 1 milliwatt. In voice transmission, the actual power will vary. A diagram showing the speech power range in *dbm* of transmitted signals at various

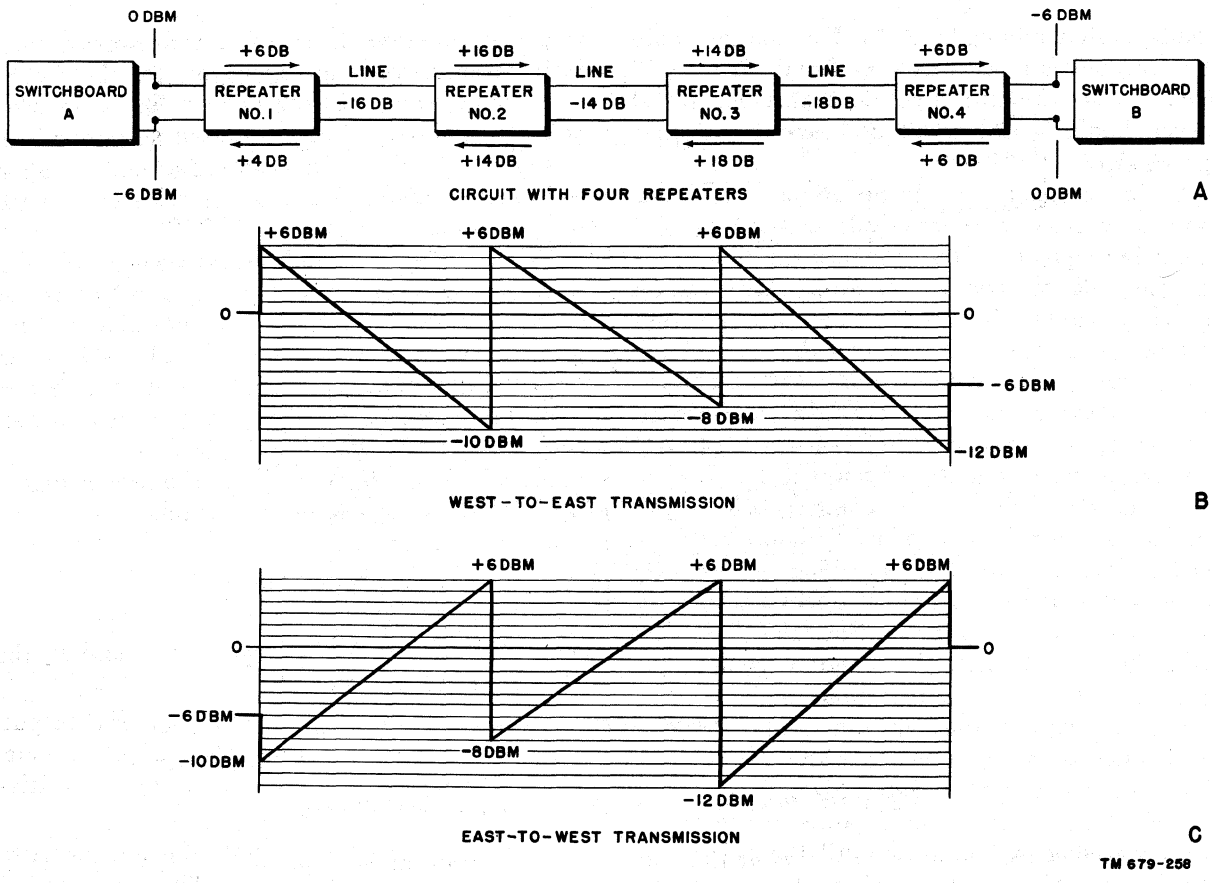


Figure 218. Two-way transmission line with power-level diagrams.

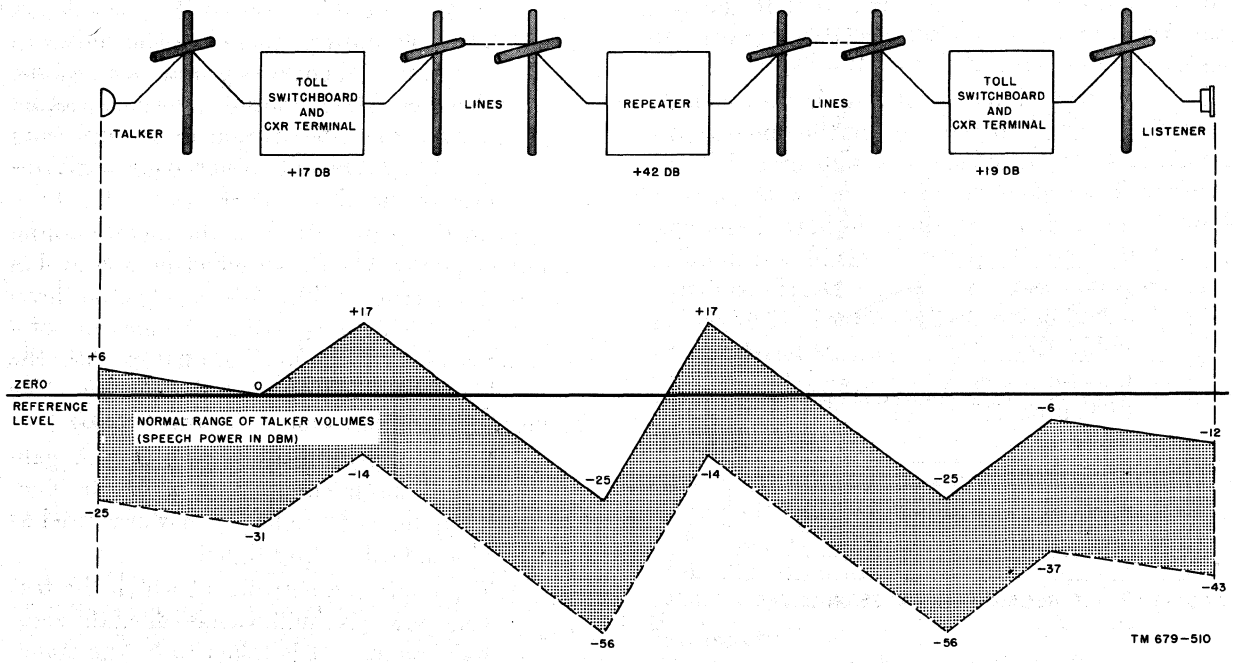


Figure 219. Normal range of talker volumes.

points, is shown in figure 219. For example, note that the talker input can vary from +6 to −25 *dbm*. It is standard practice that noise signal power should be at least 20 *db* below the weakest speech signal.

d. In telephone communications, erroneous usage of two terms commonly known as level and power is frequent. This difficulty can be cleared up when *db* or *dbm* is specified. The present tendency is to use the word "power" to mean absolute power in *dbm*. *Db* remains the ratio of the two powers. A new term, relative transmission level, is coming into use. Relative transmission level in *db* is defined as a ratio of sine wave test tone power measured at any point in the system, to the power at some other point in the system, chosen as the reference point. Figure 220 is a relative transmission diagram. A represents the power losses and gains on a line from Atlanta to Louisville, and B represents the same line from Louisville to Atlanta. The diagram enables the operators of the repeater stations to know at a glance the proper output power and repeater gains for transmission in either direction, in *db*. It should be noted in this relative transmission level diagram that, when the input power at the 0-level reference point is 1 milliwatt, all the values can also be expressed in *dbm*. That is, any value of *db* equals the same value in *dbm*.

190. Summary of Transmission Units

a. It is necessary to be able to measure the speech volume, crosstalk, and the loss caused by signal attenuation noise, on a transmission line. Such measurements are necessary for the design and maintenance of transmission-line networks. It is also necessary to have a unit in terms of which to express the measurements. The unit is called the decibel, abbreviated *db*. A number of such units is abbreviated N_{db}.

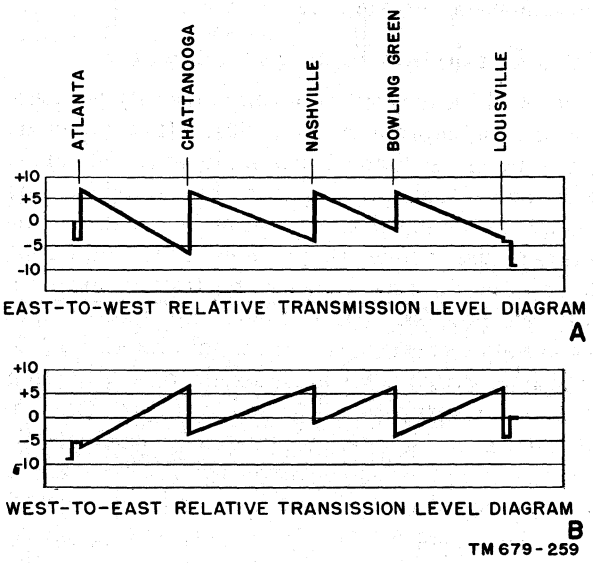


Figure 220. Relative transmission level

b. When the input and output powers of a network are in the ratio of 10 to 1, 1 bel of power change exists. When the input and output powers are in the ratio $10^{0.1}$, or 1.26, 1 decibel of power change exists. N decibels of power change exist when the input and output powers of a network are in the ratio of $10^{0.1}\,N$ times.

(1) The number of decibels of power change is found by use of the formula

$$N_{db}=10 \log \frac{P_1}{P_2}$$

where P_1 is the larger power and P_2 the smaller power.

(2) If P_1 (the larger power) is at the output, the network has caused a power gain. The gain is represented by a positive N_{db}.

(3) If P_1 is at the input, the network has caused a power loss. The loss is represented by a negative N_{db}.

(4) The decibel is a unit of power ratio, not a unit of amount of power change.

(5) The total power change in a transmission-line network is the algebraic sum of the decibels of attenuation and gain caused by a signal attenuation in the line, and the repeaters and other equipment inserted into the network.

c. The decibel can be used as the measure of an amount of power when a specified power level is used as a reference. The reference power level is 1 milliwatt. The unit used as the measure of a power level at any point as compared with the reference level of 1 milliwatt is the *dbm* (not the *db*, which is the expression of a power ratio).

(1) When measuring the power loss or gain in a speech circuit, a 1,000-cycle sine-wave tone of 1 milliwatt power is used as a standard testing signal.

(2) The point is a network at which the test tone power is 1 milliwatt is called the zero-level point. It is taken to be the transmitting switchboard.

(3) A positive number of *dbm* indicates a power level greater than 1 milliwatt.
(4) A negative number of dbm indicates a power level less than 1 milliwatt.

d. A power-level diagram (sometimes called energy-level diagram) shows the power levels throughout a network graphically.
(1) A horizontal base line represents a level 0 *dbm*.
(2) Positive values are located above the zero-*dbm* base line.
(3) Negative values are located below the zero-*dbm* base line.
(4) Power changes caused by equipment are represented by vertical lines drawn from the horizontal line representing the power at the input to the piece of equipment, to the norizontal line representing the power level at the output.
(5) Power losses resulting from attenuation along the transmission line are drawn diagonally downward.

Section II. METHODS OF TRANSMISSION MEASUREMENT

191. Need for Transmission-Measuring Set

a. Since the decibel is the expression of the ratio of two power levels—an input level and an output level—it would seem logical to measure first the power put into a circuit, then the power taken out of the circuit, and then determine the power ratio in decibels by use of the formula

$$N_{db}=10 \log \frac{P_1}{P_2}.$$

This process is slow, however, because of the mathematical calculations involved, and therefore measuring devices that read directly in decibels have been developed. These devices are of two types: the comparison type and the direct-reading type. Transmission-measuring sets of the comparison type compare the loss in a circuit with a known calibrated loss. Those of the direct-reading type measure the loss in a circuit without reference to any standard circuit.

b. Basically, the methods used in both types of measuring set are the same. A standard test tone is supplied at the sending end of a circuit and the resulting output is measured at the receiving end. The ratio of these two powers, expressed in decibels, is a measure of the loss or gain in the circuit. The magnitude of the test-tone power is small, and therefore the meter has to be very sensitive. Since d-c meters are usually more sensitive than a-c meters, they are usually preferred. Before a d-c meter can be so used, however, the a-c test power must be converted into direct current by means of a vacuum-tube or copper-oxide rectifier.

c. Theoretically, the simplest type of transmission-measuring set might contain only a source of testing power and a d-c meter calibrated in decibels; but such a simple set could not be used in practice. In the early sets the meters were not sufficiently sensitive and the amplifiers that were used to make up for the lack of sensitivity were not stable. As a result, the comparison-type measuring set came to be used. This set compares the loss or gain of equipment under test with the loss or gain of equipment that is used as a standard. In most cases, any variation caused by the instability of the amplifier would be the same for both the unknown and standard circuits, and therefore would not show up in the comparison of the losses or gains.

d. The use of the direct-reading measuring set was made possible by the development of highly stable amplifiers, copper-oxide rectifiers, and more sensitive d-c meters. Present-day measuring sets, used in the broadcast field, can measure 20 *db* below 1 milliwatt—that is, 10 microwatts—without the use of an amplifier.

192. Comparison-Type Measuring Set

a. A block diagram of a comparison-type measuring set is shown in A, figure 221. Its component parts are an oscillator which produces a 1,000-cycle test-tone signal; an amplifier that amplifies this signal; an artificial transmission line with a calibrated dial which shows how much different lengths of the line decrease the testing power; a variable line (another artificial transmission line of variable length, also with a calibrated dial); a multiple switch allowing passage of the testing power through either the artificial line, via terminals A, or the variable line, via terminals B; another amplifier; a detector for converting the a-c test signal to direct current; and a d-c meter.

b. The operation of this comparison-type measuring set can be understood by reference to the diagram in B figure 221. First, the test-tone sig-

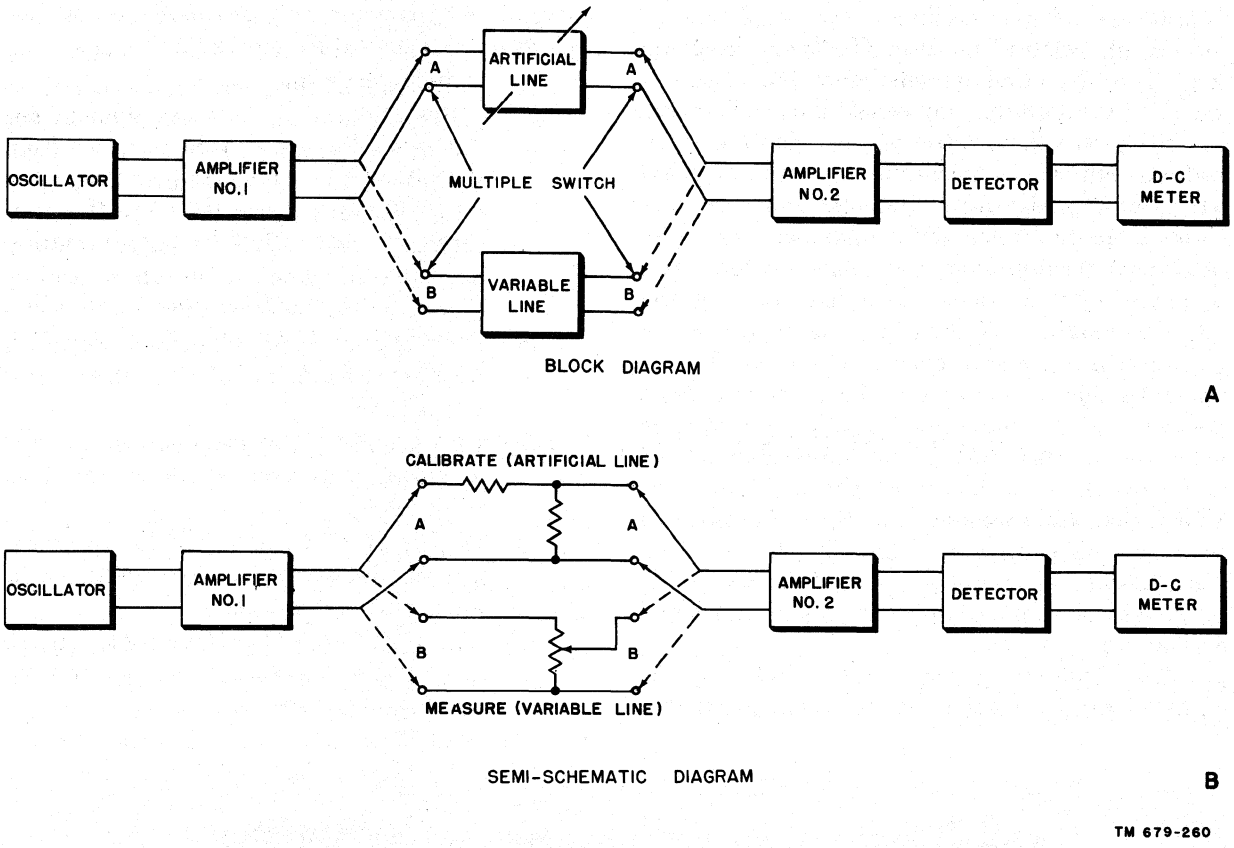


Figure 221. Diagrams of comparison-type measuring set.

nal is applied to the artificial line, represented by the resistors called Calibrate. The attenuation caused by this line is set at a fixed figure; assume it to be 20 *db*. The output of amplifier No. 2 is adjusted so that its meter needle is at midscale. The testing signal then is applied to the variable line, represented by the variable resistor called Measure. Then the attenuation caused by the variable line is adjusted so that again the meter reads midscale. The variable line now is causing the same amount of attenuation as the artificial line. The output of the amplifiers must remain fixed, and not varied, once the measurements are started.

c. The unknown line or piece of equipment then is connected in series with the variable line. The total attenuation will be greater than the attenuation resulting from the variable line alone, and the meter needle will swing away from its midscale position. The attenuation caused by the variable line now is decreased until the meter needle returns to its midscale position. The total attenuation caused by both the variable and unknown lines now is equal to the 20-*db* attenuation indicated by a midscale meter reading. The loss of the line or piece of equipment being measured is the difference between the readings of the dials of the artificial (CALIBRATE) and variable (MEASURE) lines. If the dial on the variable line indicates, for example, 11 *db* of attenuation, the line or piece of equipment being measured must be causing $20-11=9$ *db* of attenuation.

d. This method is used also to measure circuit gains. For circuit gains, the total attenuation resulting from the variable line and the circuit being measured is less than the attenuation caused by the variable line alone. The variable-line attenuation is increased, therefore, to return the meter needle to its midscale position. Suppose the variable-line dial reading is now 35 *db*. The circuit gain being sought is $35-20=15$ *db*. The unknown gain is the difference between the readings of the dials of the artificial (CALIBRATE) and variable (MEASURE) readings.

e. For making measurements, the input impedance of the artificial line is matched to the

impedances of the oscillator (or amplifier) at the sending end and the amplifier at the receiving end. But the input impedance of the unknown line or piece of equipment almost always is different from the impedance of the artificial line; therefore, there is a mismatch between the oscillator (or amplifier) and the unknown line, and between the receiver and the unknown line. The resulting reflection losses are added to the losses produced by the transmission line, causing the total attenuation as read by the meter to be greater than the actual attenuation caused by the unknown line. In practice, the reflection losses are reduced to a minimum by including variable networks in both sides of the measuring set for matching the impedances of the oscillator, unknown line, and amplifier; but the adjustment of these variable networks and of the artificial transmission line and other parts of the measuring set makes the operation cumbersome and time-consuming.

f. The number of dials and switches to be operated can be observed on the front panel of a comparison-type measuring set used for carrier-frequency measurements (fig. 222).

(1) As a result of the need for making these many adjustments, the use of this type of set has steadily decreased, whereas the use of the direct-reading measuring set has steadily increased. The DIAL–1, DIAL–2, and DIAL–3 potentiometers must be adjusted to obtain a reading of 0 *db* on the meter scale.

(2) The test-current potentiometer permits a precise adjustment of the input test power.

(3) 0–50 *db* and 0–10 *db* attenuators vary the attenuation in steps of 10 *db* and 1 *db* respectively.

(4) The ADJUST-COMPARE key provides a means for setting up two circuit paths for comparison purposes.

(5) The CALIBRATE-MEASURE switch is used for calibrating the set and for making measurements.

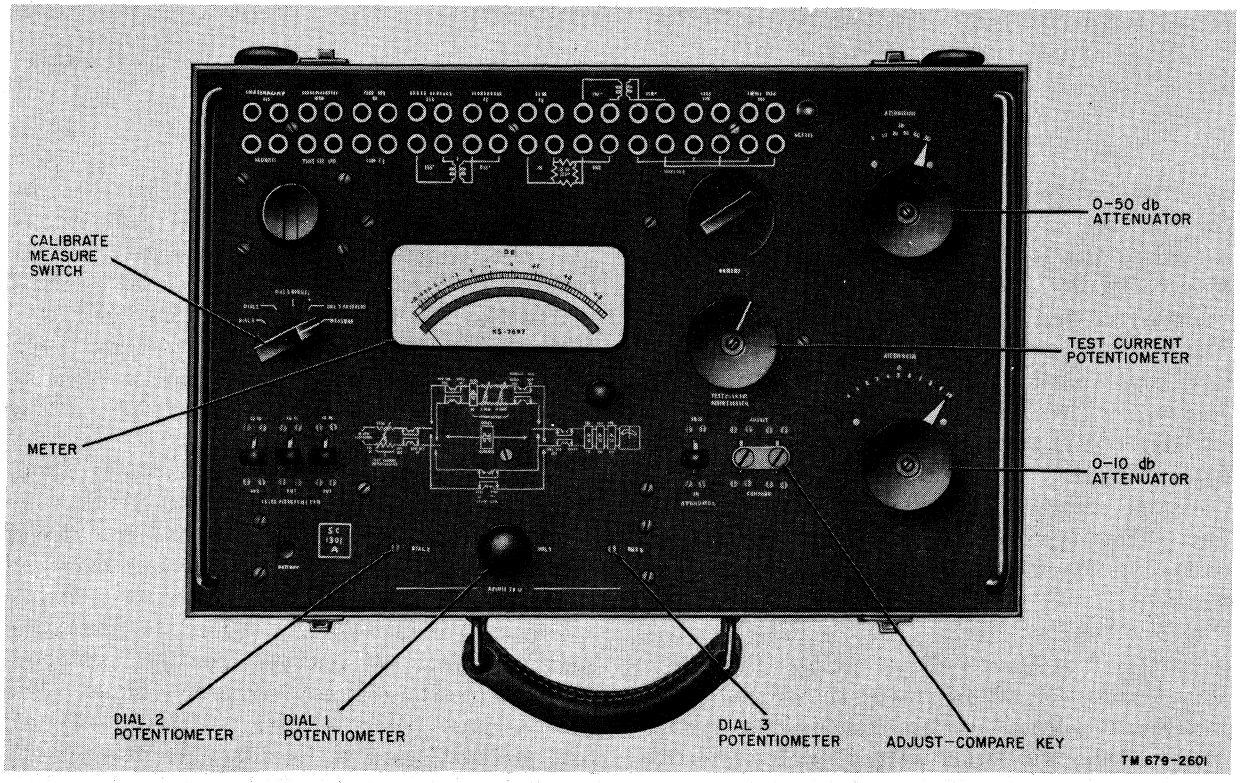


Figure 222. Comparison-type measuring set.

193. Direct-Reading Measuring Set

a. Direct-reading Measuring Set (Without Amplifier).

(1) Figure 223 is a schematic of a simple direct-reading measuring set that operates without an amplifier. A 1,000-cycle, 1-milliwatt test tone is applied to the input of a transmission line or piece of equipment, the attenuation of which is unknown. The output of the transmission line or equipment is the signal to be measured by the measuring set. A filter is inserted at the input end of the measuring set to eliminate signals with frequencies outside the range between 850 and 1,150. The signals outside this range, caused by interference induced in the network being measured, would seriously affect the amplitude of the signal transmitted by the network.

(2) The filtered signal is applied across four copper-oxide rectifiers which act as a full-wave rectifier (fig. 223). Rectifiers 1 and 4 permit flow of current through the meter during the first half-cycle of the a-c signal; rectifiers 2 and 3 permit flow during the second half-cycle. Note that the flow of current is in the same direction through the meter during both half-cycles. The a-c test signal thus has been converted into a d-c signal for use with the more sensitive d-c meter.

(3) Key-controlled attenuating pads are inserted into the measuring set to extend its range. (A pad is a resistive network used to attenuate a signal by a known amount.) A, figure 224, shows a pad disconnected from the circuit of a set. With contact A open and contact B closed, $R1$ is shorted out and $R2$ is open-circuited, and the meter reads the signal directly. In B, a pad is shown connected in the circuit of the set. With contact A closed and contact B open, the signal is applied into the series-shunt combination of $R1$ and $R2$, and the output is taken off across $R2$. This signal has been reduced in the ratio $R2$ $(R1+R2)$. Assume that $R1$ equals $R2$. The output is then one-half of the input. Without the pad an input current of 1 ma causes a full-scale deflection of the meter needle. With the pad an input current of 2 ma causes full-scale deflection. If the meter dial is scaled 0 *db* (zero attenuation) to 10 *db*, without the pad it measures a range of 0 *db* to 10 *db*; with the pad its range is 0 *db* to 20 *db*. Use of the pad doubles the signal current that can be measured by the meter, and thus doubles the range of the meter. In figure 223, the two pads give two extra scales for the meter.

(4) The impedance-improving network (fig. 223) is a variable network for improving the match between the network on which the measurement is being made, and the combination of rectifier and d-c meter which comprise the measuring set proper.

(5) The calibrating resistance serves to adjust the current flow to the meter. The meter scale is calibrated in decibels. When the sending and receiving ends are directly connected, the meter scale should read 0-*db* loss, for when so connected there is no attenuation of the test signal. If the meter does not read 0-*db* loss, the calibrating resistor is adjusted so that it does. Thereafter, in measuring an unknown attenuation, the meter pointer

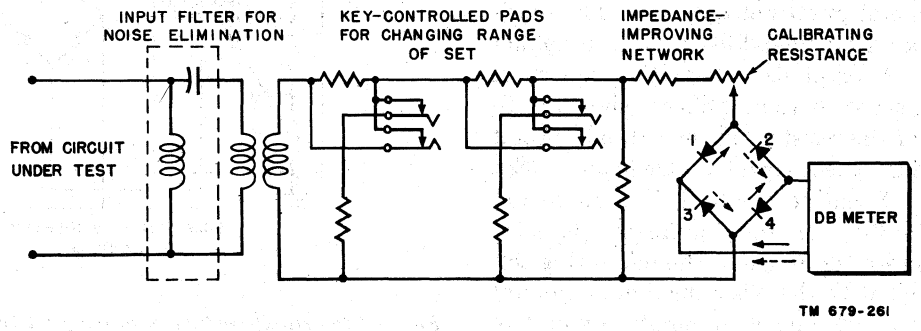


Figure 223. Direct-reading measuring set (without amplifier).

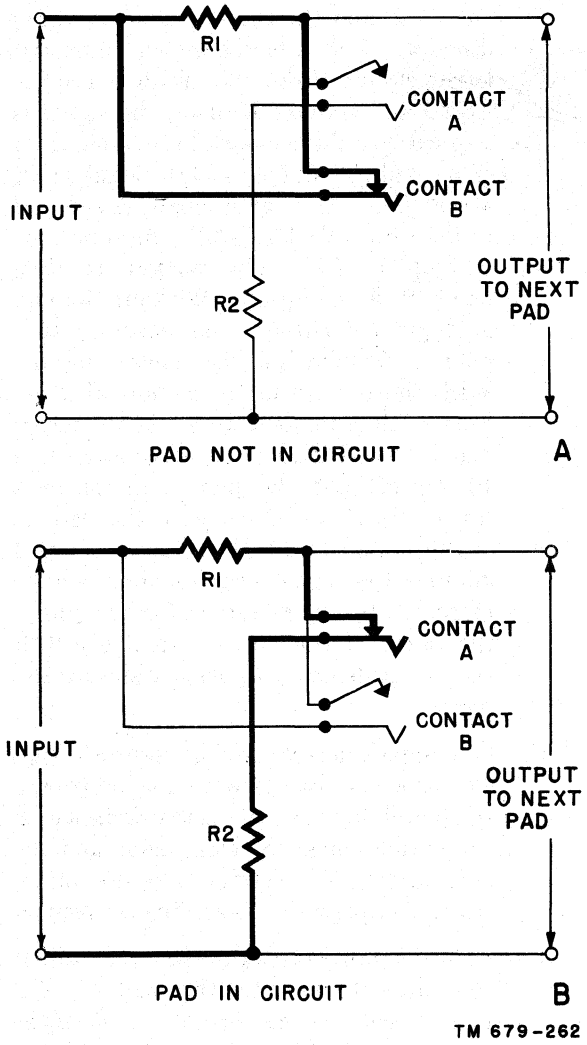


Figure 224. Use of pad to extend meter range.

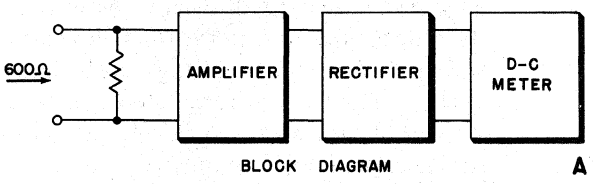
will turn in proportion to the transmission loss.

b. Direct-Reading Measuring Set (with Amplifier). When the loss to be measured is greater than 20 *db*, the simple measuring set described above is insufficiently sensitive to measure the very small amount of output current. The sensitivity is increased by the addition of an amplifier (A, fig. 225). A simplified schematic diagram of the measuring set with an amplifier is shown in B. The presence of the resistors in the cathode circuit of the amplifier causes negative feedback. (See TM 11–670.) Negative feedback is used to improve the stability of the amplifier. Selector switch A increases the range of the measuring set. With the switch at B, there is a maximum amount of negative feedback and a minimum amount of gain. With the switch at D, there is minimum feedback and maximum gain. Thus, with the switch at D, the set can measure a weaker current—a greater amount of attenuation—than when the switch is at B. The resistors are of such value as to increase the range in steps of 10 *db*.

c. Advantages of Direct-Reading Measuring Set. The front panel of a direct-reading measuring set (fig. 226) shows why this type of measuring set is simpler to operate than the comparison type. Only three adjustments need be made—Operate the ON-OFF power switch, set the range of the set on the DIAL switch, and use the calibrating rheostat to cause the meter to read 0-*db* loss when the sending and receiving ends are directly connected. The direct-reading measuring set is less expensive than the comparison type, but it is not as precise, because of changes in the meter, the rectifier, and the amplifier which occur with aging.

194. Decibel Meters

a. Db meters of both comparison-type and direct-reading measuring sets usually have a d-c ammeter or voltmeter with the scale calibrated to read decibels directly. The scales of two types of *db* meters are illustrated in figure 227. On both



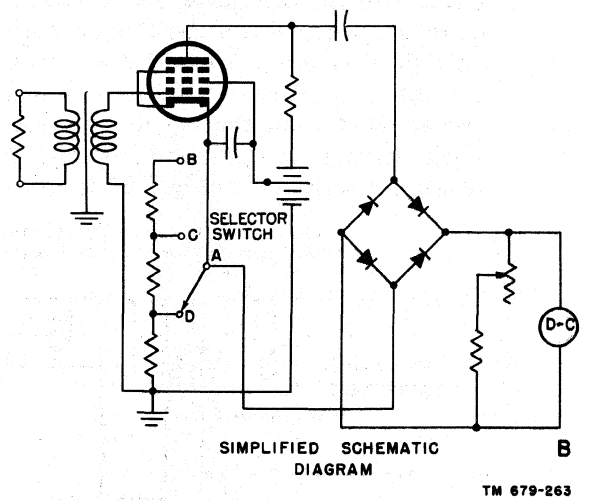


Figure 225. Direct-reading measuring set (with amplifier).

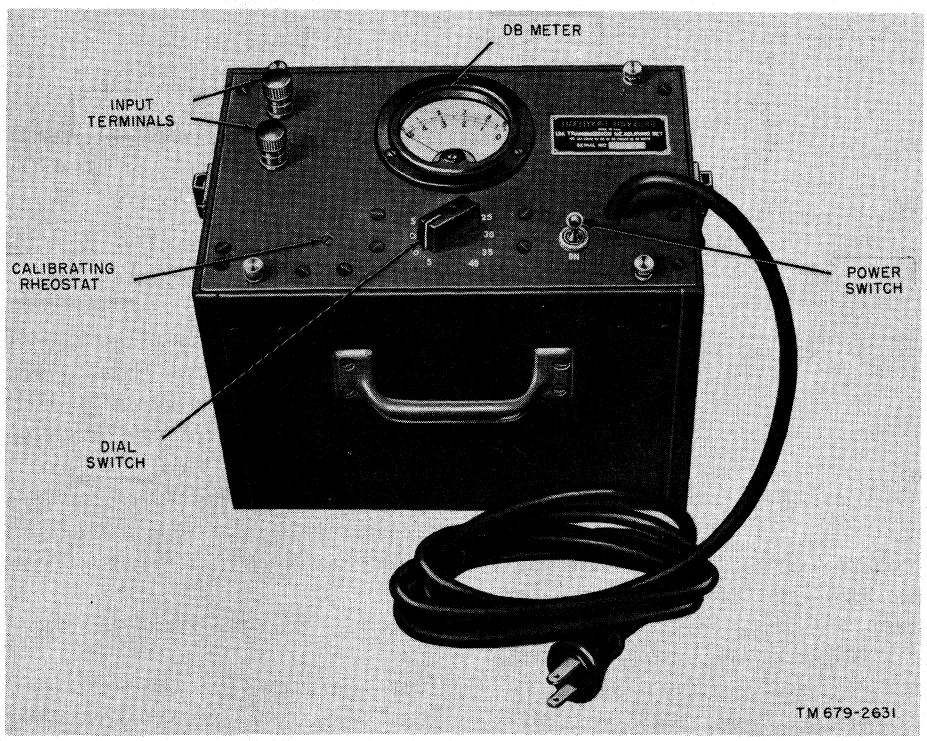


Figure 226. Direct-reading measuring set.

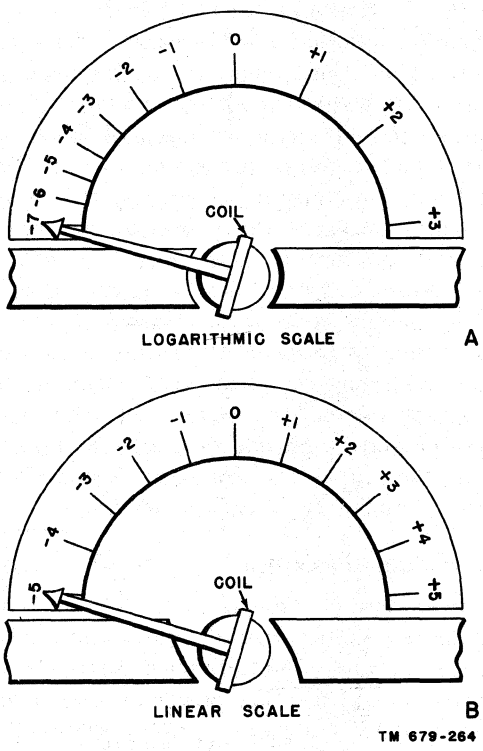


Figure 227. Decibel meters.

types of meters, 0 *db* is found near the center because these meters are designed to read attenuation above and below the 0-level reference. Thus a reading of 0 *db* is produced by the amount of current corresponding to the fixed input power of the test signal, usually 1 milliwatt.

b. The scale of A, figure 227, is nonlinear. An ordinary ammeter has a linear scale; equal changes of current produce equal deflections of the indicator needle. This results from the fact that the coil bearing the indicator needle rotates in a uniform magnetic field. But (as indicated in table IV (par. 185), and fig. 228) the logarithmic relationship between number of decibels and change of power causes the number of decibels of change to increase rapidly for small changes of current, and slowly for large changes of current. The nonlinear meter scale shown in A, figure 227, is logarithmic to correspond to this relationship—the larger decibel readings are spaced far apart, the smaller ones close together.

c. The linear scale shown in B, figure 227, is usually more desirable than a nonlinear scale because it permits readings of equal accuracy over the entire scale. On the nonlinear scale, the read-

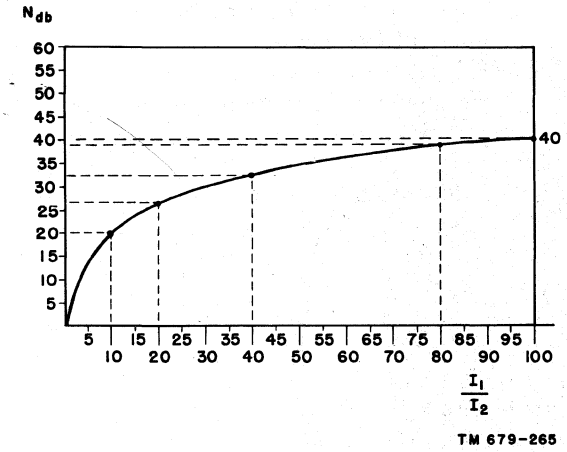


Figure 228. Logarithmic relationship between current and N_{db}.

ings at the low *db* end of the scale are less accurate than at the high end, because the scale divisions are smaller. The linear *db* scale is obtained by varying the width of the air gap in which the coil bearing the indicator needle rotates. The poles of the permanent magnet are somewhat irregular, causing the air gap in which the rotating coil moves to vary. The magnetic field, therefore, is weaker at the high-current end of the scale than at the low. The number of decibels of change and the magnetic field both being logarithmic functions of the current in the meter, the number of decibels varies directly as the magnetic field, and the result is a linear scale.

195. Speech Measurement

a. Making transmission-line measurements of sine-wave signals of constant amplitude (usually 1 milliwatt of power) and of single frequency (usually 1,000 cycles) is a relatively simple problem. At times, however, it is necessary to measure speech-signal levels. This is not a simple problem, because speech signals have no single frequency. Speech signals are composed of many components, each of a different frequency, and usually all of different amplitude (fig. 229). Also, the speech signal is nonperiodic—the succession of values of the signal do not repeat themselves at regular intervals. Thus, there is no simple or constant relationship among the peak, average, and rms (root mean square) values of the signal. A meter that gives steady readings when used on a single-frequency constant-amplitude sine wave would behave in a very erratic manner when used to measure a speech signal. If the meter were a peak-reading one, the needle would not remain

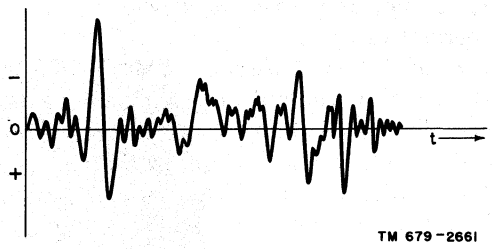


Figure 229. Speech signal.

stationary, because the peak values of the speech signal vary from instant to instant. Such erratic variation would occur also if the meter were reading either rms or rectified average values of a speech signal.

b. A sine wave has three distinguishing characteristics—peak value or amplitude, frequency, and initial phase angle. In the equation of a sine wave,

$$e = A \sin (2\pi ft + \phi),$$

these characteristics are designated respectively by A, f, and ϕ. However, although they may differ from sine wave to sine wave, the manner in which the instantaneous values vary from instant to instant is the same for all sine waves; that is, all their waveshapes are the same. As a result, there is a fixed relationship among the peak, average, and *rms* values of all sine waves. A meter calibrated to read any specific one of these values, within a certain frequency range, will read in the same fashion for any other sine wave. A meter designed to measure the power in any one sine wave will measure the power in any other sine wave. Note, however, that a speech signal (fig. 229) completely lacks these consistent sine-wave features such as constant amplitude, constant frequency, and constant phase angle, and the waveshapes are seldom the same. Measurement of the power and amplitude of a speech signal cannot be accomplished by the straightforward methods applied to pure sine waves.

196. Volume Indicator

a. Definition of Volume Indicator. A volume indicator is a meter devised to measure the power (volume) in speech signals with reference to an arbitrary standard. The power in speech signals is measured by the meter across a 600-ohm load resistance. The variations in power are expressed on the meter face in terms of decibels above and below the arbitrary standard. The meter does not read the actual instantaneous power level;

rather, it indicates average power values over short periods of time.

b. Uses of Volume Indicator. Volume indicators have three major uses in the measurement of speech signals. They are employed in the prevention of overloads on recording devices; in the monitoring of radio broadcasts transmitted to networks over telephone transmission lines; and in the measurement of attenuation of speech signals in circuits carrying speech traffic.

c. Units Used in Volume Indicator.

(1) Although the volume level is expressed as a number of decibels above or below an arbitrary standard, it is not customary when dealing with volume levels to speak in terms of decibels. In order to distinguish between the power in a sine wave (expressed in decibels) and the power in a speech signal, a new unit, the volume unit, or *vu*, has been adopted to express the latter. Thus, instead of speaking of N_{db} above or below 0 volume level, it is customary to speak of the number of *vu* above or below the reference volume level.

(2) The volume indicator is calibrated to read 0 *vu* when the indicator is connected across a 600-ohm resistance in which is flowing 1 milliwatt of sine-wave power at 1,000 cycles. In use, the indicator reads the volume level in terms of *vu* above or below a reference level which is defined as that level of program (speech signal) which causes a volume indicator when calibrated and used in the accepted way, to read 0 *vu*. Two points must be emphasized. Both the *dbm* and the *vu* as units correspond to a power ratio of 10^{0} [1]; the *dbm* is associated with steady, single-frequency signals, whereas the *vu* is associated only with signals containing many components of different frequency and different amplitude.

d. Special Characteristics of Volume Indicator. The meters used in making measurements on sine-wave signals are inadequate for making measurements on speech signals. As a result, the meter in a volume indicator must have special characteristics in order to fulfill the special requirements.

(1) The meter is calibrated across a 600-ohm resistor. This is the input resistance of many circuits used in communication networks, and many of the measurements to be made will be across resistances of this value.

(2) The meter characteristics are such that the meter variations are correlated with the response of the ear. If they are not, the meter readings will give no true indication of what the ear hears.

(3) The ear does not respond to the maximum volume of a sound unless the sound is present for an appreciable time, at least 0.2 second. For this reason, the meter is damped physically, so that a sound of any intensity causes it to read 99 percent of total scale deflection in 0.3 second. A sound of equal intensity but shorter duration causes a smaller meter deflection (a lower reading). In this way, the meter is caused to read an average value of speech-signal power, not an instantaneous value.

(4) The ear does not respond to changes of sound volume in a linear manner. When the power level of a speech signal is halved, that does not mean that the resulting signal appears half as loud to the ear. The ear responds to changes of sound volume in a logarithmic manner. Since the decibel system is logarithmic, the meter scale is calibrated in volume units, which are simply decibels renamed for application to speech-signal measurement.

(5) The sudden application of a single frequency signal that gives a steady-state reading of 0 level causes the meter pointer to overswing by 1 to 1.5 percent. Such overswing, caused by underdamping, makes for a less "jittery" meter.

(6) The response of the meter should be uniform over a frequency range from 35 cycles to 10 kc.

e. Other Possible Types of Volume Indicator. The volume indicator responds to *rms* values of a speech signal. With special meter types, the rectified average value (average amplitude) and peak value of such signals may also be read.

(1) A meter with a uniform scale, preceded by a rectifier, will read average values of speech amplitude directly, but average values have no real meaning in speech-signal measurements. The *rms* value is of far greater significance, because of the constant relationship between *rms* value

and power. (P equals I^2R, where I is the *rms* value of the current.)

(2) Other meter circuits may be devised to measure peak signal values. Since overloads on recording devices are caused by the peak values of speech signals, such a meter might appear useful; but tests show that peak measurements have no advantage over *rms* values. A change in the waveshape of the speech signal may seriously change its peak value without significantly affecting its *rms* value. For this reason, peak measurements are not made.

197. Theory of Operation of Volume Indicator

The volume indicator correlates sound intensity with the amplitude of the voltage which the sound produces in the transmitting instrument and transmission line.

a. The volume indicator is a kind of voltmeter, calibrated in *db* across a 600-ohm load (for reasons stated in par. 196). It is applied across the circuit like a voltmeter, as shown in figure 230. Since the intensity of the sound causing the generator output voltage is directly proportional to the output voltage, the volume indicator, by measuring the voltage amplitude, gives an indication of the volume of the sound.

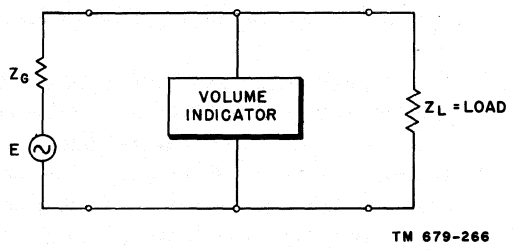


Figure 230. Circuit of volume indicator.

b. The lowest scale of the volume indicator is calibrated at the +4-*vu* level, not the 0-*vu* level. With 0 as the lowest meter scale, and a 0-*vu* value corresponding to 1 milliwatt in a load of 600 ohms, the voltage across the line would be

$$E=\sqrt{PR}$$

$$E=\sqrt{(.001)(600)}$$

$$E=.775 \text{ volt } (rms).$$

Thus, for the indicator to read 0 *vu*, the speech voltage would have an *rms* value of 0.775 volt. This value causes a relatively small meter deflection and severely limits the range of the meter. With +4 *vu* as the lowest meter scale, however, the power in the 600-ohm load is +4 *db* above 1 milliwatt, so that the power in the load is 2.513 milliwatts and the voltage across the load is

$$E=1.228 \text{ volts.}$$

This larger voltage causes a greater needle deflection and gives a greater meter range. However, +4 *vu* now must be added to the observed values in order to read true *vu* (or true *db* with respect to the fixed level—that is, *dbm*).

c. The voltages just computed are *rms* values for single-frequency signals. For speech signals, such values are composites of all the *rms* voltages that make up the complex signal.

d. When the indicator is connected across a load resistor of value other than 600 ohms, the meter reading does not indicate the true volume of the signal. For example, a +4-*vu* reading across a 500-ohm resistor corresponds to an *rms* signal of 1.228 volts. The power in the resistor would then be 3.015 milliwatts. This is higher than the value of 2.513 milliwatts which would be dissipated in the 600-ohm load. Hence, the meter would not be reading true with respect to the chosen reference. A correction factor must be used, computed from the formula

$$N_{db}=10 \log \frac{Z_0}{Z_1},$$

where Z_0 is the 600-ohm calibrating resistance and Z_1 is the actual load impedance. With the 500-ohm load the correction factor is

$$N_{db}=10 \log \frac{600}{500}$$

$$N_{db}=10 \log (1.2)$$

$$N_{db}=+.792.$$

Thus, the corrected volume level is +4+.792= +4.792. A meter reading of 4.792 *db* above the reference level of 1 milliwatt corresponds to 3.015 milliwatts. This is the actual power in the 500-ohm resistor, as was calculated above. Note that if Z_1 is greater than Z_0, the correction factor is

negative, making the corrected reading less than that observed on the meter face.

198. Circuit of Volume-Indicator

a. Figure 230 is a diagram of a simplified version of the volume-indicator circuit. The meter has an internal impedance of 3,900 ohms; sufficient resistance is added in series (3,600 ohms) to give the instrument an overall input impedence of 7,500 ohms. When the volume indicator is connected across a 600-ohm load, the relatively high 7,500-ohm input impedance of the indicator draws minimum power from the load, so that the volume of the speech signal in the load remains substantially unchanged. When the selector switch on the front panel of a volume indicator is set to the +4-*vu* scale (fig. 231), the meter pointer is deflected to the 0 *vu*, or 100 mark, when the *rms* value of the voltage across the load is 1.228 volts. The true *vu* level under this condition is +4 *vu*.

(1) For reading higher speech-power levels, the +4-*vu* scale is inadequate. On the panel of the volume indicator (fig. 231), there are provisions for 11 other levels above the +4-*vu* level. The position value is added to that which the meter reads. (This correction holds for the +4-*vu* scale as well as for the other scales.) For example, if the switch at the +6 position and the meter reads 0 *vu*, the true reading is 0+6 *vu*, or +6 *vu;* if the meter reads −2 *vu*, the true reading is −2 +6, or +4 *vu*. The additional levels are made possible by an attenuator T pad.

(2) The attenuator T pad (A, fig. 232) is a variable resistor network which is used in a line to cause a desired voltage drop, at the same time providing an impedance match in both directions. Its three resistances are connected in an electrical T. The three arrows joined by the dashed line indicate that the resistances are ganged and connected to one control. When the shaft of the control is rotated, the resistances are varied simultaneously to effect automatically the desired voltage drop, at the same time maintaining the impedance match.

(3) As used with the *vu* meter, the T pad is made variable to drop the voltage applied to the *vu* meter in order to make the meter read 0 *vu* when the signal volume equals the switch setting. The T pad matches in such a fashion that no matter which setting is chosen, the resistance at point H, looking toward the meter, is always 3,900 ohms. This can be seen by the settings of the resistances as shown in figure 232, B and C. In B, the pad is shown in a counterclockwise position. The signal voltage is shorted at the top of resistor 2, but the resistance at both the input and the output still remain the same at 3,900 ohms. In C, the pad is shown in a clockwise position. Resistances 1 and 3 are shorted out completely and the signal voltage passes through without attenuation. Resistance 2, in this case, maintains the impedance constant since all of it is in

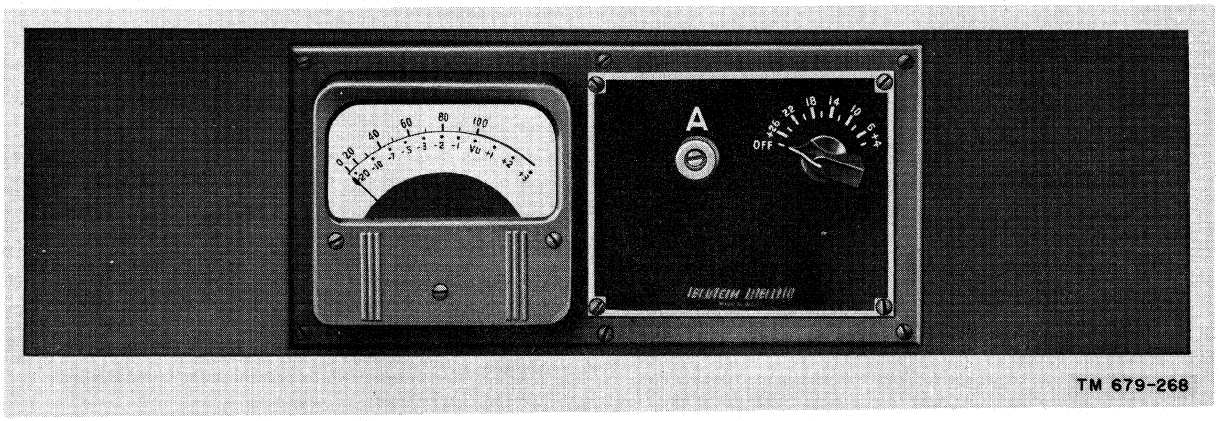


Figure 231. Panel of volume indicator.

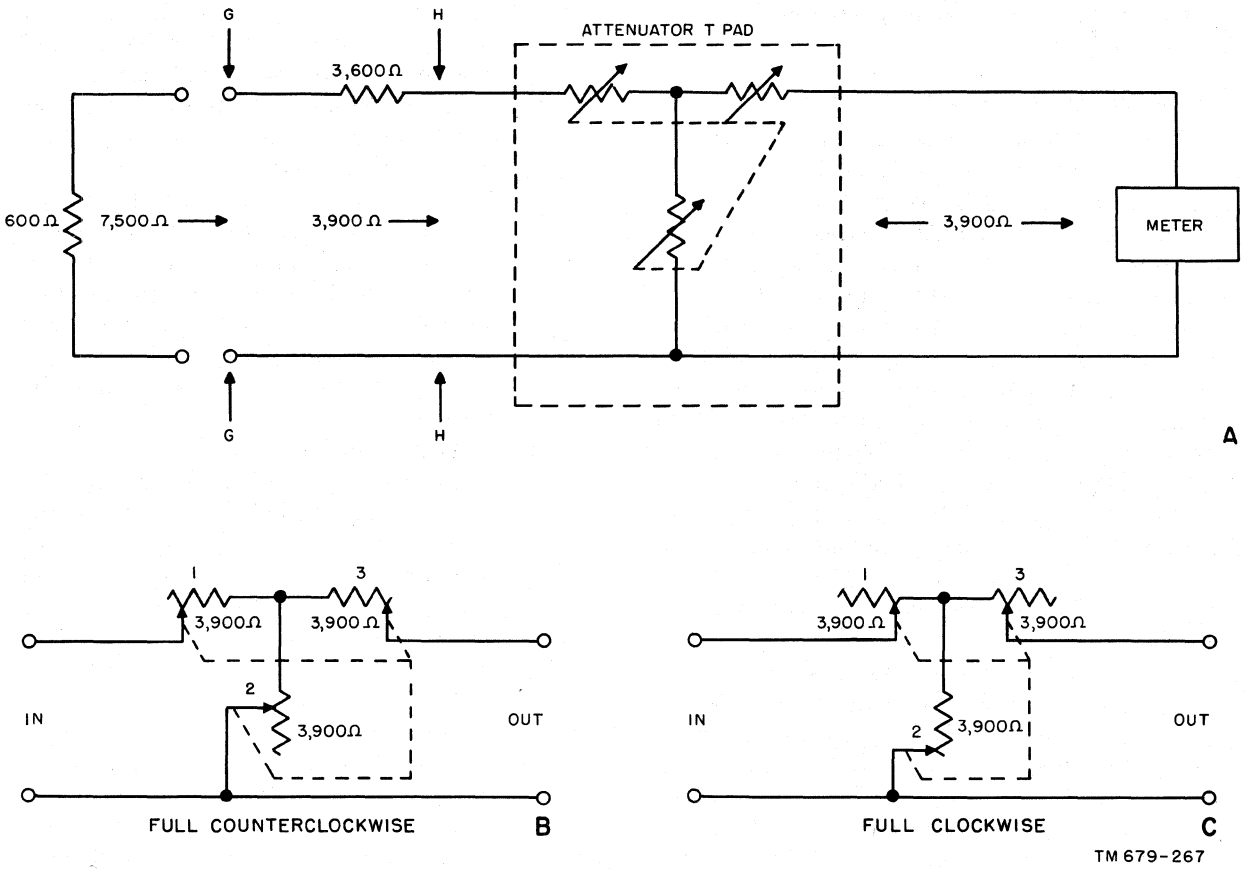


Figure 232. Simple diagram of standard volume indicator.

the circuit. Thus, the voltage applied to the meter may be attenuated without changing the apparent resistance looking toward the meter from points H. When the fixed resistor of 3,600 ohms is added between points G and H, the overall resistance of the *vu* meter, at its terminals, is a constant 7,500 ohms.

(4) The function of the T pad may be illustrated by noting that for the meter to read 0 *vu* when the switch is in the 10 position, the power in the 600-ohm load (across which the meter terminals are connected) must be 10 milliwatts (10 *db* above 1 milliwatt). Then, the voltage across the load must be

$$E=\sqrt{PR}=\sqrt{(.010)(600)}$$

$$E=2.450 \text{ volts.}$$

However, only 1.228 volts is required to deflect the meter to 0 *vu*. The difference in voltage, 2.450—1.228, or 1.222 volts appears across the pad. Similarly, for the meter to read 0 *vu* across 600 ohms when the switch is in the +22 position, 8.528 volts appears across the pad, and 1.228 volts across the meter.

b. On the +4-*vu* scale, 1.228 volts across the 600-ohm load deflects the meter needle to the 0-*vu* point. On the +22-*vu* scale there are 9.756 volts across the 600-ohm load, but the attenuator reduces the voltage sufficiently to deflect the needle to the 0-*vu* point. In this way, additional scales are made possible. If the scale selector is on the +4-*vu* scale, but the voltage across the 600-ohm load is less than 1.228 volts, the meter needle is deflected to a point below the 0-*vu* mark. This indicates that the speech-signal power in the load is less than 4 *db* above 1 milliwatt. Thus, the meter is used to measure variations about the power level which causes the meter to register 0 *vu*. In figure 231, the meter itself reads variations from —20 *vu* to +3 *vu*. To these direct readings are added the value of the switch position.

c. No 0-*vu* scale is provided on the volume indi-

cator because provision for such a scale would limit the range of the instrument. If such a scale were included, the 0-*vu* point would be about half-scale on the meter. By using +4-*vu* as the lowest range, however, it becomes possible to read directly −20 *vu*—that is, a true reading of −16 *vu* when the +4 of the switch position is added to the observed value. This follows because the 0 point has been moved up the scale to about two-thirds full-scale.

199. Use and Calibration of Volume Indicator

a. The meter faces in figures 231 and 233 are different versions of the same type. The meter face of figure 231 accentuates the 0-to-100 scale, whereas the face of figure 233 accentuates the *vu* scale. The difference is a matter of emphasis, not of type. If use of the volume indicator makes it more desirable to emphasize the 0-to-100 scale, the scale on figure 66 is used; if the *vu* scale is to be emphasized, the scale on figure 233 is used. Both scales are read in the same manner, however. When the selector switch on the front panel is set at +10 *vu*, a meter reading of 0 *vu*, or 100, represents a volume level 10 *vu* above 0-*vu* level. The voltage across the load must now be 2.45 volts, and the voltage drop across the T pad must be of such value that the needle is deflected to the 0-*vu* mark. As a result, any variation in the volume level is now about the +10-*vu* level, rather than about the 0-*vu* level. If the meter reading is such that the needle is deflected to 18 *vu* below 0 *vu* with the switch in the +10-*vu* position, the volume level is 18 *vu* below the +10-*vu* level, or the actual reading is +10 *vu*−18 *vu*=−8 *vu*. Similarly, if the meter reading is 3 *vu* above 0 *vu* with the switch on the +4-*vu* scale, the volume level is 7 *vu* above 0-*vu* level. This means that there are 5 milliwatts of power in the speech signal.

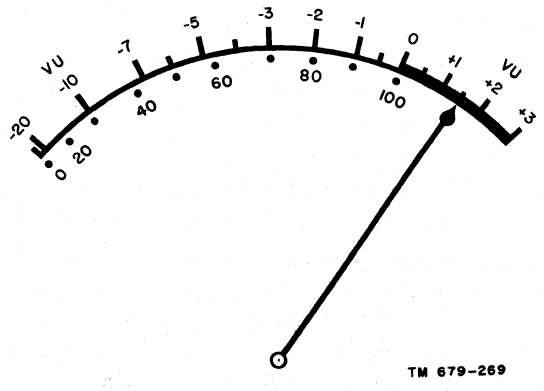


Figure 233. Second form of vu meter.

b. The calibration of the meter for operation on any scale requires two simple operations. First the selector switch on the front panel of the volume indicator is set to the +4-*vu* range. With the meter connected across a 600-ohm load which is dissipating 4 milliwatts of sine-wave power at a frequency of 1,000 cycles, the screw adjustment A (fig. 231) is turned until the meter reads 0 *vu*. The meter now reads correctly on all ranges, the switch-position value being added to that read from the meter face.

200. Correlation of *Db, Dbm, Vu*

a. Each of the units for expressing power levels (the *db*, the *dbm*, and the *vu*) has its particular sphere of usefulness; only in exceptional cases do their functions overlap. The *db* is used as a measure of the ratio of any two power levels of a steady single-frequency signal. The power level of a steady, single-frequency signal may be expressed as a certain number of *dbm*, where the power level is a number of decibels above or below the specific reference level of 1 milliwatt. The *vu* indicates the volume level of a dynamic speech signal (as distinguished from a steady, single-frequency signal) as being a certain number of *vu* above or below a specific reference level—the level of a speech signal which causes the volume indicator to read 0 *vu*.

b. The *db* and *dbm* functions overlap when one of the power levels in the *db* ratio is 1 milliwatt. Actually, under such conditions, the meter is reading *dbm*, since the *dbm* reference level is so specified.

c. There is no numerical correlation between *dbm* and *vu* reading, because *dbm* uses 1 milliwatt as a reference level, and the volume indicator selects any of several other reference levels, the lowest being +4 *vu* above 1 milliwatt. A further difference between *dbm* and *vu* is that the *vu* indicates the level of a dynamic signal containing many components of different frequency and amplitude, whereas the *dbm* indicates the level of a signal of single frequency and constant amplitude. *Vu* can be read only on a volume indicator, for only this type of meter has the necessary properties. *Dbm* can be read on any proper type of meter, including the volume indicator (provided, in the case of the volume indicator, the scale value is added to the reading). When a volume indicator is used to read *dbm*, the readings are the same;

for instance, 10 *vu* may be interpreted as 10 *dbm*. However, this is the exceptional case where the volume indicator is used on a single frequency signal. Normally, it is used on complex signals, containing components of many different frequencies.

d. The decibel is a measure of power ratio (consequently of power level above or below some reference level) for sine-wave signals. The *vu* reading of a volume indicator is a measure of power level above or below the level designated by the selector switch—for complex speech signals. If the volume indicator is used with a sinusoidal (single-frequency) input, its readings may be interpreted without correction as *db* above or below the level, in *dbm*, indicated by the scale selected.

201. Crosstalk

a. Measurements of signal attentuation on a transmission line, and the decibel system of expressing transmission line losses and the gain caused by a repeater (pars. 191–200), are needed to determine the amount of attentuation on a line, so that repeater amplifiers may be properly designed to restore the signal to its original amplitude. Measurements are needed also in dealing with crosstalk, another form of signal deterioration. Crosstalk changes the shape of the transmitted signal. It may cause speech signals to suffer a loss of intelligibility, and cause teletype signals to make errors on a teletypewriter.

b. Crosstalk is said to exist in a communication system when the signal transmitted in one circuit or channel can be detected in another circuit or channel. The first of these circuits, the one over which the signal is transmitted, is called the disturbing circuit. The second circuit, the one in which the undesired signal has been induced by crosstalk coupling, is called the disturbed circuit.

c. Coupling between two pairs of transmission lines may exist either between two pairs of telephone lines or between a telephone line and a power line. Three types of signals can be developed in one of the telephone lines as a result of such coupling. One, interference from a powerline, is usually in the form of noise, and appears as a humming or buzzing in a receiver at the end of the telephone line. If the interference is between two telephone lines, it appears in the receiver as either an intelligible signal, one that can be understood, or as an unintelligible signal, one that cannot be understood and sounds like noise. Crosstalk is said to exist in a common communication system only when the interference is between two telephone lines, and the interference causes the signal transmitted in the disturbing circuit to be intelligible in the disturbed circuit. An unintelligible signal in the disturbed circuit is considered a form of noise, although sometimes it is called unintelligible crosstalk.

d. Only intelligible interference is considered crosstalk. Unintelligible crosstalk is considered noise, because its effect on the disturbed circuit is similar to the effect of interference from a power line. However, in considering methods of eliminating interference between pairs of transmission lines, no distinction should be made between intelligible crosstalk, unintelligible crosstalk, and power hum noise. All are induced in the same way, and all are eliminated by the same methods.

202. Crosstalk Coupling

a. Crosstalk coupling between a disturbing and a disturbed circuit is defined as the ratio of the induced power in the disturbed circuit to the transmitted power in the disturbing circuit. Crosstalk coupling is usually called crosstalk-loss and is expressed in decibels by the use of the formula

$$N_{db}=10 \log \frac{P_{\text{crosstalk}}}{P_{\text{signal}}}.$$

As an example, let the crosstalk power and the signal power be in the ratio of 1 to 1,000,000; the crosstalk loss in decibels is

$$N_{db}=10 \log \frac{P_{\text{crosstalk}}}{P_{\text{signal}}}=10 \log \frac{1}{1{,}000{,}000}$$

$$=10 \log 10^{-6}=-60 \text{ } db.$$

b. The major causes of crosstalk (par. 33) in brief review are—

(1) Electrostatic coupling exists between pairs of lines when the electric field of one pair causes a voltage to be developed in the second pair.

(2) Electromagnetic coupling exists between two pairs of lines when the magnetic field of one pair induces a voltage in the second pair.

(3) Conductive coupling may exist if there is leakage between two pairs of conductors as a result of faulty insulation between them.

(4) Coupling caused by a resistive unbalance exists when there is a difference in the resistances of the two wires of a pair; as a result, the currents induced in the two wires are unequal, even if the induced voltages are equal.

c. Of the cases of crosstalk-coupling in *b* above, the two first are the result of the positions of transmission-line pairs relative to one another; the third and fourth are the result of the characteristics of the lines. An additional source of interference and crosstalk coupling is the unbalance that exists when the pieces of equipment inserted into the two wires of a circuit are not sufficiently alike in impedance at the transmission frequencies. This type of unbalance is similar to the resistive unbalance described in *b*(4) above. The cause does not result from a dissimilarity in the resistances of the wires, however; unbalance in this case is caused by the dissimilarity in the impedances of pieces of equipment inserted into the lines. As a result of this unbalance, the currents induced in the two wires are unequal even if equal voltages are induced in the two wires by electrostatic or electromagnetic coupling. When the induced voltages are equal, no crosstalk results; but the impedance unbalance caused by the equipment causes the currents in the two conductors to be unequal, and in this way it is a cause of crosstalk.

d. There is still another manner in which a specific piece of equipment—a repeater, for example—can cause interference when inserted into a line. A repeater is an amplifier inserted into the line to make up for the signal attenuation in the wires of the line (par. 2). This type of amplifier should operate in a linear manner; that is, on the linear portion of the transfer characteristic (i_b, e_c) curve of the tube in the amplifier, since such operation prevents distortion of the transmitted signal. If the repeater amplifier operates in a nonlinear manner, however, signals not present in the input appear in the output. (See TM 11–670.) These signals are distributed over a wide frequency range and are heard as a humming noise in the receiver. Thus, nonlinear operation of a repeater results in distortion of the transmitted signal and interference to the desired signal on the line.

203. Types of Crosstalk

a. A signal transmitted over the disturbing circuit may be heard at either end of the disturbed circuit. Figure 234 illustrates crosstalk coupling between the disturbing circuit and the near and far ends of the disturbed circuit. The interference is called near-end crosstalk, as shown in A, when the crosstalk is heard on the disturbed circuit at the same end of the line at which the talker is energizing the disturbing circuit. The interference is called far-end cross-talk when the crosstalk is heard on the disturbed circuit at the opposite end of the line, as in B. If the same band of frequencies is transmitted in both directions over the disturbing and disturbed circuits, both types of crosstalk may occur. This may happen in circuits carrying voice frequencies, and in some carrier-frequency systems. With many carrier systems, the same frequency is not sent in both directions on the circuits. The outgoing signal on the disturbing circuit is purposely at a different frequency from the returning signal on the disturbed circuit. This eliminates near-end crosstalk coupling; but the outgoing signal on the disturbed circuit, and as a result far-end crosstalk coupling exists.

b. Experiments indicate that a listener has difficulty in distinguishing between the main talker and a crosstalker unless the speech signal he receives from the main talker is at least 13 *db* higher than that from the crosstalker. The ratio of received speech signal—main talker to crosstalker—is call the signal-to-crosstalk ratio. This ratio determines how much crosstalk coupling can be permitted in a circuit of known attenuation. Figure 235 represents a case of near-end crosstalk. The speech volume of the main talker is −5 *vu*. Attenuation reduces the signal to −35 *vu* at the listener. The speech volume of the crosstalker is also −5 *vu*. A crosstalk coupling loss of 43 *db* reduces the crosstalk-speech volume to −48 *vu* at the listener. The signal-to-crosstalk ratio is 13 *db* (the difference between −35 *vu* and −48 *vu*). Thus, when the transmission line causes 30 *db* of signal attenuation, 43 *db* of crosstalk loss is the least that can be tolerated. An even greater crosstalk loss (less crosstalk coupling) is desirable. The figure of 43 *db* is based on the assumption that the speech volumes of the main talker and crosstalker are equal. But if the speech volume of the crosstalker is greater than that of the main talker, the crosstalk loss must be greater than 43 *db* to produce the desired minimum signal-to-crosstalk ratio of 13 *db*.

c. The presence of repeaters on the line, and their effect on the crosstalk signal, must be con-

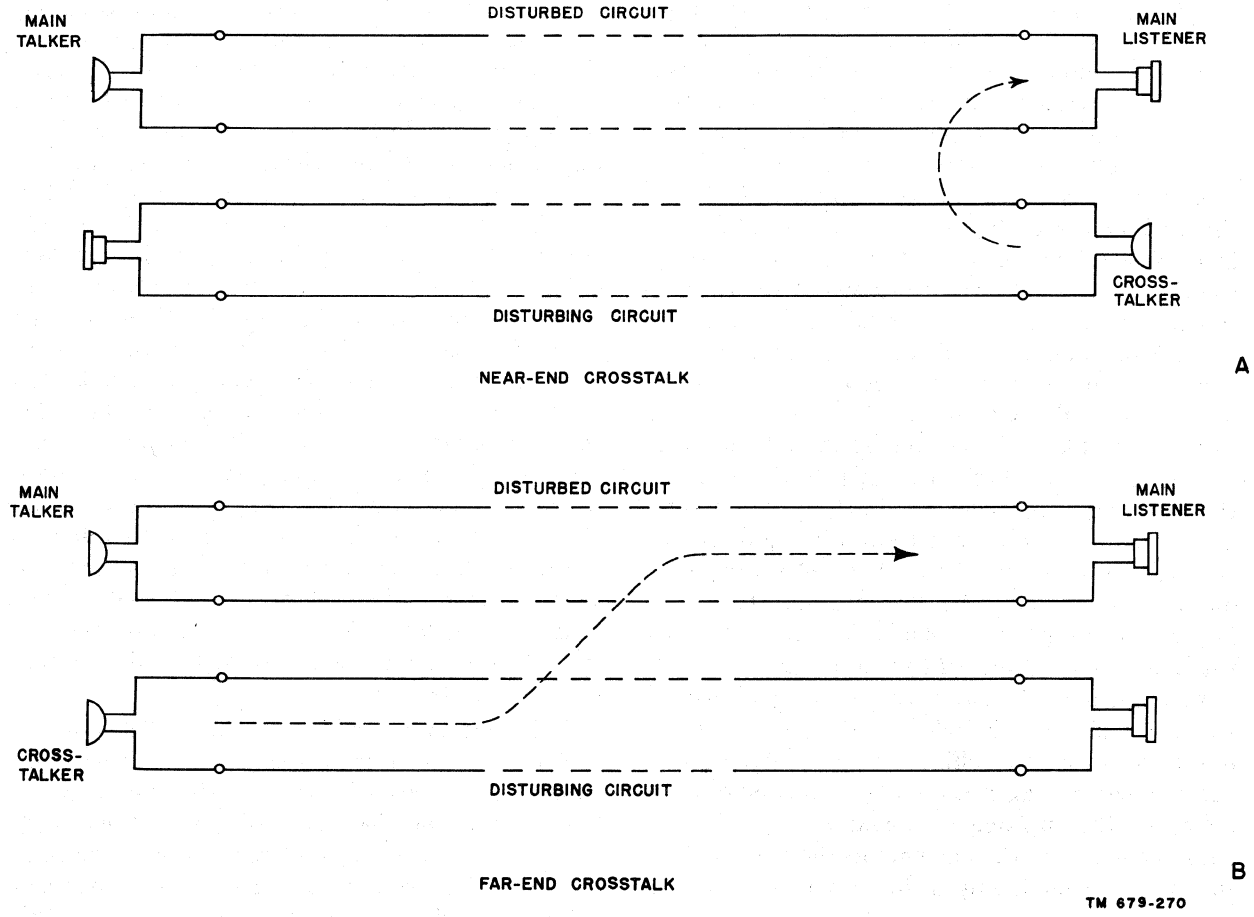


Figure 234. Near-end and far-end crosstalk.

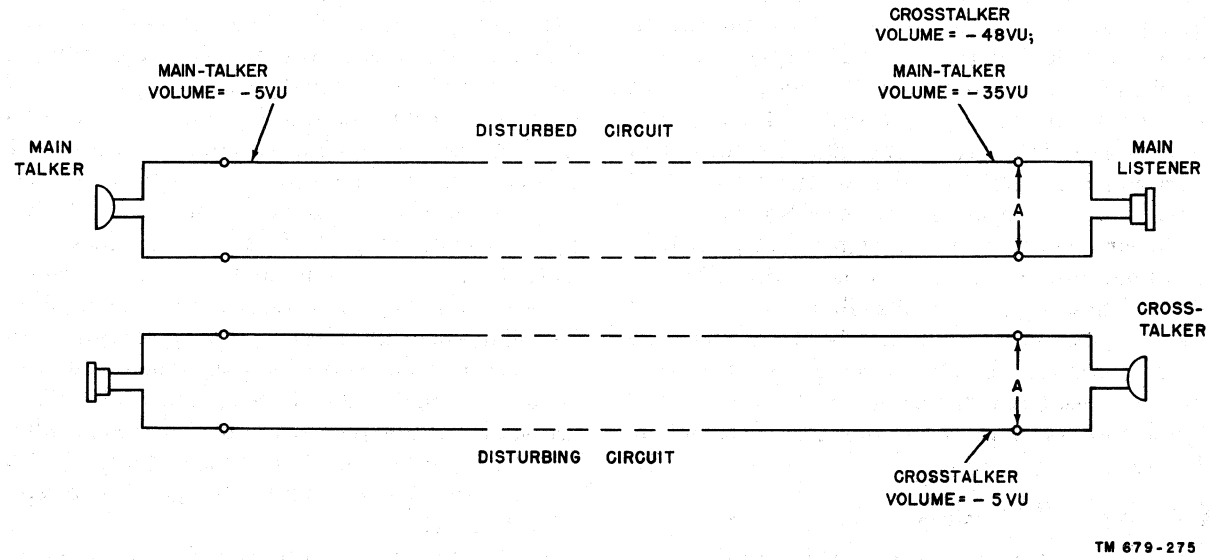


Figure 235. Crosstalk volume.

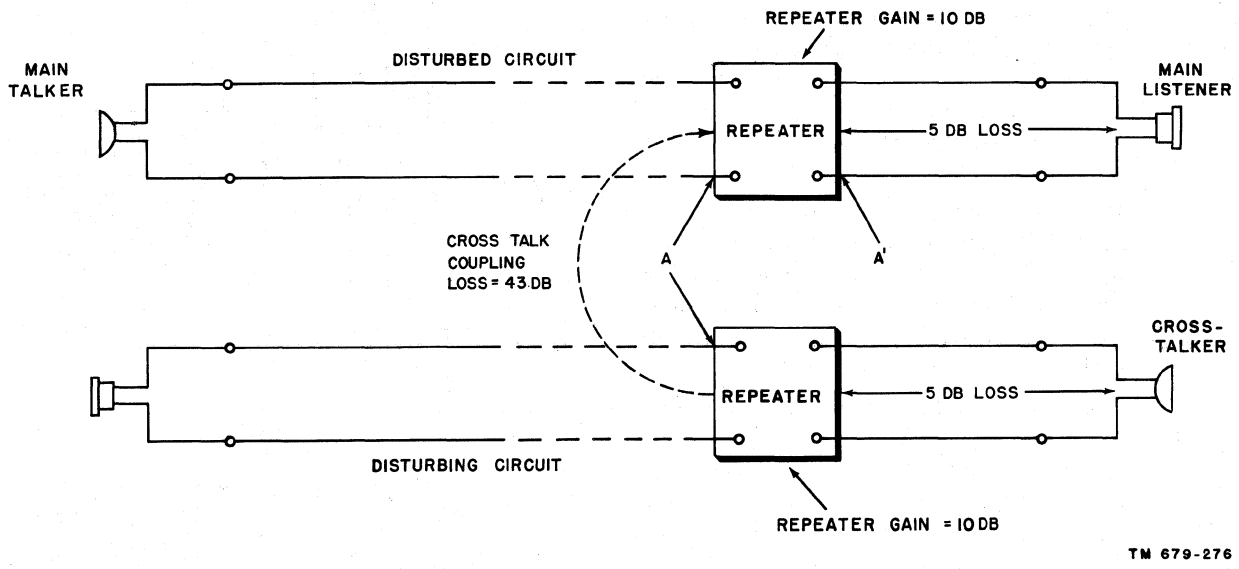


Figure 236. Loss and gain in crosstalk path.

sidered in determining how much crosstalk coupling can be tolerated. Figure 236 shows two transmission line pairs on each of which there is a repeater.

(1) The transmitted signal from the crosstalker undergoes a 5-*db* loss because of the line attenuation, and then a 10-*db* gain because of the repeater. Assume that the crosstalk-coupling loss at A is 43 *db*, as in the example in paragraph 202*a*. In the disturbed circuit, the crosstalk signal again undergoes a gain of 10 *db* and a loss of 5 *db*. The total change in the crosstalk signal is $-5+10-43+10-5=-33$ *db*. If the speech volume of the crosstalker is -5 *vu*, the change causes the crosstalk volume to be -38 *vu* at the main listener.

(2) Assume, as in the example in *b* above, that the speech volume of the main talker is also -5 *vu*, and that the net effect of line attenuation and repeater gain has reduced the signal to -35 *vu* at the listener. The signal-to-crosstalk ratio, then, is $-35-(-38)=+3$ *db*, which is 10 *db* less than the $+13$-*db* ratio required for the signal to be intelligible.

(3) In the example in paragraph 202*a*, in which there were no repeaters to consider, 43 *db* of crosstalk-coupling loss could be tolerated to produce the required 13-*db* signal-to-crosstalk ratio. In the present example, using the same speech and crosstalk volumes, a crosstalk coupling loss of only 43 *db* causes the signal to be completely unintelligible. The gain which the crosstalk signal undergoes, called crosstalk amplification, makes it necessary that there be far less crosstalk coupling (greater crosstalk-coupling loss). With 15 *db* of crosstalk amplification, as in the present example, crosstalk coupling loss must be at least 53 *db*, instead of the 43 *db* that can be tolerated when there is no crosstalk amplification.

d. For purposes of computation, it is sometimes convenient to consider that the crosstalk amplification is the same at all points on the line. It is highly improbable that such a condition ever will exist, but by making it appear that crosstalk amplification is the same at all points, such differences as do exist may be eliminated from consideration, and total crosstalk loss at any point may be computed with reference to a common base. Crosstalk loss computed in this way is referred to as equal-level crosstalk loss.

(1) At A in the disturbing circuit (fig. 236) the crosstalk signal is at a level $-5+10=+5$ *db* above the signal level at the crosstalker. This is also the crosstalk amplification at A in the disturbed circuit. A, where the crosstalk coupling takes place, will be taken as the reference point in finding the equal-level crosstalk

loss, even though the signal is raised another 10 *db* between points A and A′.

(2) At A, the crosstalk coupling loss is —43 *db*, and the crosstalk amplification is +5 *db*, so that the net crosstalk loss is —38 *db*.

(3) At some other point, B (not shown in figure 236), let the crosstalk amplification be +8 *db*, 3 *db* more than at A, and the crosstalk coupling loss —39 *db*, making the net crosstalk loss —39+8=—31 *db* at B.

(4) To place this point on an equal level with A, it must be made to appear that the crosstalk amplification is the same at both points. At the same time, the net crosstalk loss at B must remain unchanged. This is done by taking the difference between the crosstalk amplification values, +3 *db*, and adding it algebraically to the crosstalk coupling loss at point B: —39+3=—36 *db*. This is the equal-level crosstalk-coupling loss at point B. Added to the reference point gain of +5 *db*, it gives the actual net crosstalk loss at point B as required: —36+5=—31 *db*.

(5) The total crosstalk loss between the crosstalker and the listener is simply the sum of the equal-level crosstalk-coupling loss at the point in question and the net loss of the disturbed circuit between that point and the listener. Thus, if crosstalk loss exists at two points such as A and B, the total crosstalk loss can be computed by first obtaining the total crosstalk loss for each crosstalk path and then by means of available charts (such as in fig. 5–82, TM 11–486), the entire crosstalk loss can be computed.

204. Measurement of Crosstalk

a. Figure 237 is the schematic of a crosstalk-measuring set. The components of the measuring set are an oscillator as an a-c source; a calibrated shunt attenuator which attenuates the signal from the a-c source; terminals to which the disturbing and disturbed lines can be connected; a switch by means of which the signal from the a-c source can be connected to either the disturbing line or the calibrated shunt; and a telephone receiver (it could be a meter) in which the signal from the

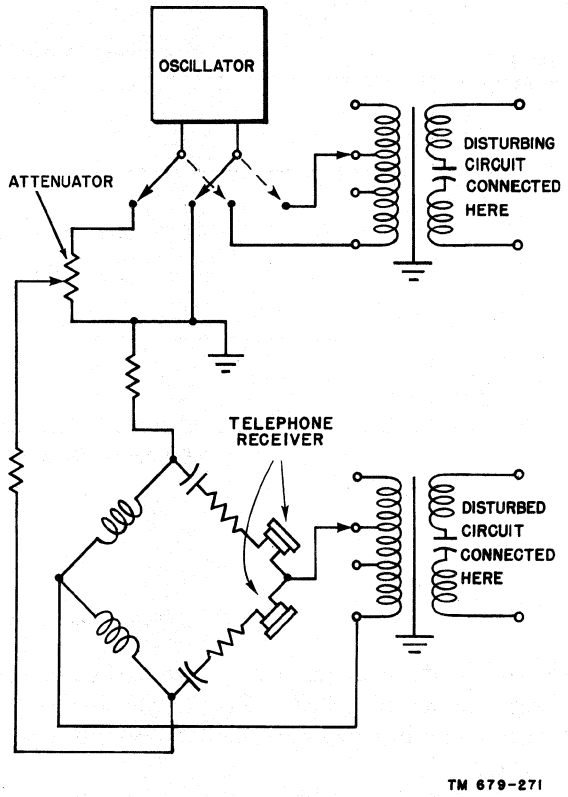


Figure 237. Diagram of crosstalk-measuring set.

calibrated shunt can be compared with the signal from the disturbed line.

b. When measuring crosstalk coupling, the output of the oscillator is connected first to the disturbing line by means of the switch (fig. 237). Crosstalk coupling causes the signal to appear on the disturbed circuit where the signal is monitored in the telephone receiver. The switch then is thrown so that the testing tone is connected to the calibrated shunt. The signal is attenuated by the shunt. The switch is thrown from one position to the other; that is, from disturbing line to shunt. The shunt is varied until the attenuation of the test tone causes the signal heard in the receiver on the shunt position to be of the same amplitude as the signal heard in the receiver when the switch is connected to the disturbing line. Signals of equal loudness are connected to the disturbing line and the calibrated shunt. When the signals heard in the receiver are equally loud, the signal current in the disturbed line is equal to the signal current from the shunt. The reading on the calibrated shunt, therefore, can be used as the measure of crosstalk coupling. If a meter is used in place of the receiver headphones, the

meter reads the rectified average value of the signal. The meter usually is mounted on the front panel of the measuring set. The entire unit is similar in appearance to the noise-measuring set illustrated in figure 241. The front panel of a typical crosstalk measuring set includes a calibrated potentiometer to provide various scales; calibrate-adjust screw which varies the current in the meter; the meter; and a headphone jack.

c. This crosstalk-measuring set can be used for measuring both near-end and far-end crosstalk. Figure 238 illustrates the methods used in making these measurements. When measuring near-end crosstalk, the test tone is connected to the end of the disturbing line adjacent to the end of the disturbed line at which the measurement is being made. When measuring far-end crosstalk, the test tone is connected to the end of the disturbing line farthest from the end of the disturbed line at which the measurement is being made.

205. Noise

When coupling exists between a telephone line and a power line, or between two telephone lines, the power hum or unintelligible crosstalk induced in the telephone line appears as noise in the receiver at the end of the line. However, noise can de-

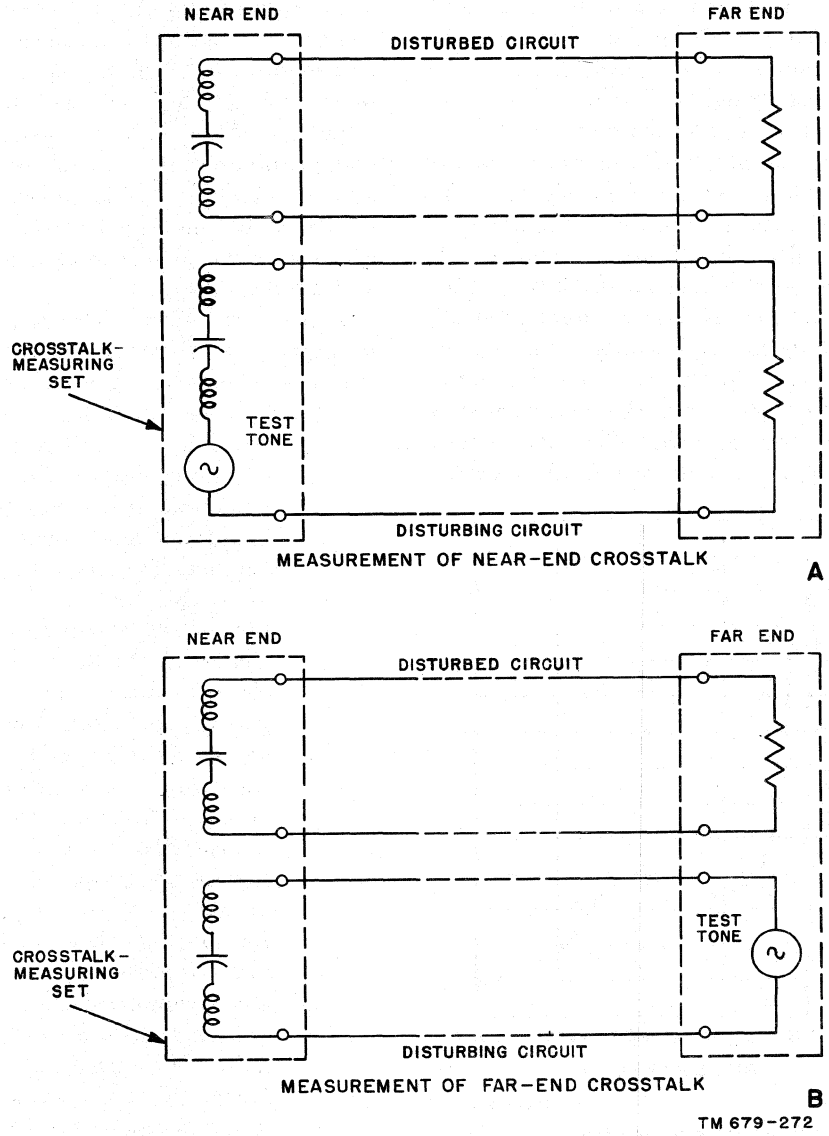


Figure 238. Crosstalk measurements.

velop in a communication circuit by means other than crosstalk coupling.

a. It should be realized that the noise heard in a receiver is the result of a voltage or current developed in the line, or in some piece of equipment inserted into the line. Just as the speech signal transmitted over the line is a voltage and current wave, so the noise signal that is developed on the line is a voltage and current wave. The noise is an unwanted signal that interferes with the speech signal.

b. There is no exact way of defining noise other than to say that it is a collection of sounds that have a disturbing effect on the ear. There usually is no fixed characteristic in terms of which noise may be defined. Noise may have any frequency. There is no constant relationship between the peak and rms values of the noise signal. The energy in the noise signal is distributed over the frequency range in a random manner.

c. The disturbing effect of the noise usually depends on the volume of the noise signal. The frequency of the noise is also very important in determining its disturbing effect. Figure 239 indicates the manner in which the disturbing effect varies with frequency. Noises of equal intensity are most disturbing in the frequency range from 800 to 1,200 cycles. In the very low frequency range, 500 cycles or less, and in the high-frequency range, 4,000 cycles or more, the effect of the noise is negligible as compared with the effect of a noise of equal intensity in the 800-to-1,200-cycle frequency range. The effect of the noise in the low- and high-frequency ranges is not negligible if the noise has a large amplitude.

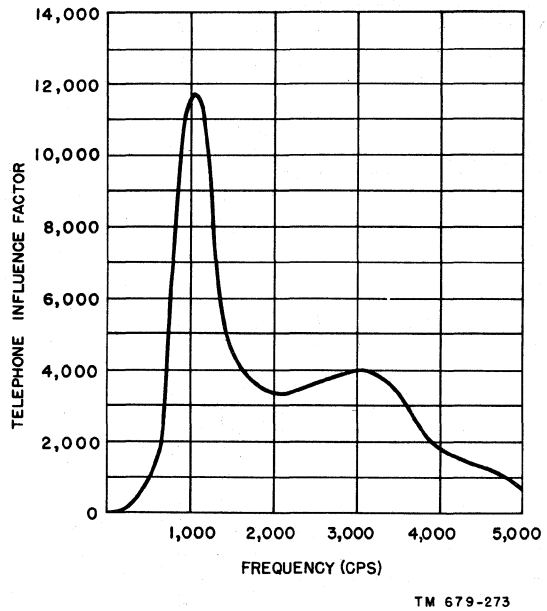


Figure 239. Interference effects of various frequencies of equal intensity.

206. Types of Noise Originating within Circuit

Although noise cannot be defined exactly, among the different types that can be recognized are impulse noise, random noise, power-hum noise, and static noise.

a. Impulse Noise. Impulse noise consists of a large number of small sinusoidal waves occurring at regular, closely spaced intervals of frequency, and possessing an in-phase relationship with one another. Impulse noise is usually in the form of a short-duration, high-current discharge. The component parts of impulse noise contain an extended series of closely spaced harmonics all in phase (since they are generated by a single impulse). Most impulse noises result from man-made equipment such as relays, motors, elevators, and automobile spark plugs. Because of the regularity of the impulse noise pattern, this type of noise sometimes is analyzed in terms of peak and average power.

b. Random Noise. Random noise is caused principally by thermal agitation and tube noise, and it is evident in a receiver as a hiss.

(1) *Thermal agitation,* one source of random noise, is the random motion of electrons caused by heat. The free electrons in a conductor are never stationary but are being moved about continuously by thermal forces resulting from the heating of the conductor. The changes of position of the electrons cause a continuous change in the distribution of the electric charges within the conductor, thereby generating small potential differences across its terminals. Alone, each free electron is the source of a separate noise voltage. Together, the electrons produce a noise characterized by many sine-wave signals of completely haphazard distribution in both frequency and phase. Because of this, such noise is called random noise.

(2) The vacuum tubes of repeater amplifiers are a second source of noise. This tube noise is caused by the shot effect. Normally, it is assumed that the current in a tube under d-c conditions is constant at all times. Actually, it is only the

average current over a period of time that is constant. From instant to instant, there is a random variation in the electron stream arriving at the plate of the tube. As a result, a random variation is produced in the output voltage of the plate-load circuit. This random voltage appears at the receiver as a random noise.

c. Power-Hum Noise. Power-hum noise may appear in a communication circuit by means other than crosstalk coupling between the communication line and a power line. As a rule, an a-c voltage is used at two points in the circuit of a repeater amplifier. The filaments of the tubes are heated by power from an a-c source, and the d-c voltage needed in the amplifier is derived from a power supply which operates from an a-c source. Interelectrode coupling between filament and cathode results in a 60-cycle modulation of the electron stream at its source—the cathode—and a resultant hum in the receiver. Insufficient filtering in the d-c supply also introduces a-c hum via the high-voltage lines. It is not the 60-cycle signal itself that is heard at the receiver. The disturbing effect of a 60-cycle signal is negligible (fig. 239); it is the harmonics of this signal that produce the disturbing noise.

d. Static Noise. This type of noise is an erratic combination of impulse and random noise. It is not inherent in the communication circuit, nor is it due to coupling between the communication circuit and some other circuit. Rather, it is manifested as an impulse or spark of high intensity. Usually it is caused by natural phenomena, such as lightning flashes.

207. Measurement of Noise

a. The intelligibility of speech signals may be determined at any one frequency by the intensity or loudness of noise on a telephone line. At some other frequency, a noise signal of the same intensity may render the speech signal even less intelligible. At a third frequency, the deterioration of the speech signal because of the noise may be negligibly small. The differences in the effect of the noise on the speech signal are caused by the relationship between the disturbing effect of the noise and its frequency. A noise-measuring set, therefore, must be capable of distinguishing the relative interfering effect of noise signals of different frequency. The readings on the meter cannot be a measure of the intensity of the noise alone; they must take into account the frequency of the noise signal. Only in this way can the significance of the noise interference be determined.

b. Noises in electrical circuits are measured on a decibel scale. Since the power in a noise signal is small compared with the usual signal power, it would be inconvenient to use 1 milliwatt as the reference noise level; therefore, electrical-circuit noises are referred to a reference power level of 1 micromicrowatt. This reference noise level is written 0 *db RN*, indicating 0+*db* above reference noise *RN*. The unit of power, 1 micromicrowatt, is equal to 10^{-12} watts. When compared to 1 milliwatt,

$$\frac{1 \text{ micromicrowatt}}{1 \text{ milliwatt}} = \frac{10^{-12}}{10^{-3}} = 10^{-9}.$$

The reference noise level is then

$$0 \text{ } db \text{ } RN = 10 \log \frac{1 \text{ micromicrowatt}}{1 \text{ milliwatt}} = -90 \text{ } db.$$

That is, 0 *db RN*, the reference noise level, is −90 *dbm*, or 90 *db* below 1 milliwatt. A noise meter is calibrated to read 0 *db RN* for a reference power of 1 micromicrowatt at a frequency of 1,000 cycles.

c. The relative interfering effect of a noise signal is determined by the intensity and the frequency of the noise, whether the noise results from crosstalk coupling or is developed in the communication circuit itself. One more factor that must be taken into consideration is that a noise signal that appears to be strong on one line because the speech signal is weak may be a weak signal on another line on which the speech signal is strong. Thus, it is the ratio of the speech and noise signals that is important. This ratio is called the signal-to-noise ratio (sometimes the speech-to-noise ratio). In telephone work, the ratio of signal and noise powers usually is expressed in decibels. It has been found that when noise is present, and the noise is either typical power induction, or uniformly distributed over the voice-frequency range, a signal-to-noise ratio of 6 *db* is about as poor as can be tolerated; further, a ratio of at least 13 *db* should be used as the basis for laying out a telephone circuit. If the ratio goes below 6 *db*, the noise causes an almost complete loss of intelligibility of the speech signal. Noise on a teletypewriter signal causes printing errors.

208. Requirements of Noise-Measuring Equipment

a. The meter in a noise-measuring set must be sensitive to a wide range of power levels. It must be capable of measuring the 0-*db RN* power level. Since sometimes the noise level may be only 6 *db* below the speech level, this places the noise level somewhere near the 0-*dbm* level; thus, the meter must be sensitive to a range of powers extending from 1 micromicrowatt to 1 milliwatt.

b. The dynamic characteristics of the meter must be similar to those of the meter in a volume indicator. Both meters must simulate the response of the ear. In doing so, the meter measures the actual interference effect of the noise.

(1) The meter is damped so that the needle deflection corresponding to a specific noise volume reaches its maximum value only if the noise lasts at least .2 second. If the same noise is of a duration less than .2 second, the deflection will be less than maximum. In this way, the meter is caused to read an average value of the noise signal, rather than an instantaneous value.

(2) The copper-oxide rectifier used in the meter is a square-law detector. The amplitude of the output of this type of rectifier is proportional to the square root of the sum of the squares of the amplitudes of the signals at the rectifier input. The noise being measured is composed of many small signals spread over a wide frequency range. The square-law characteristics of the copper-oxide rectifier cause the component noise signals to produce a resultant reading analogous to the manner in which the ear sums up component noise signals. This causes the response of the meter to be similar to that of the ear.

(3) The meter is calibrated in decibels. The logarithmic scale of the meter simulates the logarithmic response of the ear to changes of sound intensity.

c. The noise-measuring set must include a circuit not present in the volume indicator. This is the frequency-weighting circuit, which makes it possible to correlate the frequency of a noise signal and the interfering effect of the noise signal as read by the meter.

(1) Figure 239 shows that a sound of either very low or very high frequency would not be loud to the ear because of the relative insensitivity of the ear at these frequencies. A given noise containing low-frequency components of 100 cycles would not sound loud, whereas a noise of the same intensity, but containing frequency components of 1,000 cycles would seem loud. The frequency-weighting network causes the noise signal at anyone frequency to be attenuated by an amount inversely proportional to the interfering effect of a noise signal of that particular frequency. Thus, at a frequency of about 1,000 cycles, at which the relative interfering effect is maximum, the noise signal should have maximum amplitude. But at 3,000 cycles, at which frequency the relative interfering effect is only about one third as great, the amplitude of the noise signal should be only about one-third of maximum.

(2) The ear differentiates naturally between noises of different frequency. This results in the relative interfering effect versus frequency relationship, illustrated in figure 239. The meter alone is incapable of such differentiation; but the frequency-weighting network, by properly attenuating the noise signals, causes the noise signals coming to the meter to have an amplitude-versus-frequency characteristic similar to that drawn for the ear. It is in this manner that the meter in a noise-measuring set simulates the amplitude response characteristic of the ear.

209. Noise-Measuring Set

a. Figure 240 shows that the component parts of a noise-measuring set are an impedance-matching circuit, a frequency-weighting network, an attenuator potentiometer, an amplifier, a copper-oxide rectifier, and a d-c noise meter.

(1) The impedance-matching circuit is at the input to the measuring set. By means of it, the input impedance can be matched

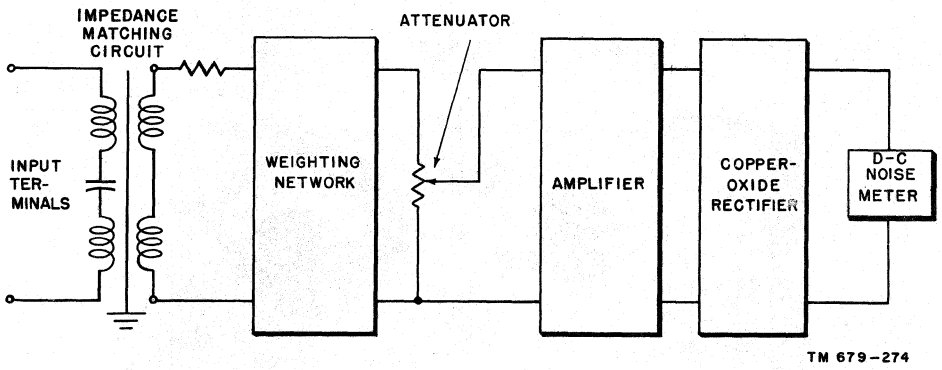


Figure 240. Diagram of noise-measuring set.

to the impedance of the circuit to which the set is to be connected.

(2) There is usually more than one frequency-weighting network that may be used with a measuring set, depending on where the measurement is being made, and the speech level. For instance, on a telephone line that transmits only the voice-frequency range, it is unnecessary to determine the disturbing effect of frequencies above 3,000 cycles; but on a telephone line carrying a radio program for a network, the disturbing effect on music of frequencies above 3,000 cycles is important. Furthermore, the ear responds in different ways to sounds of different intensities. For loud sounds, the response of the ear is much more uniform than it is for relatively weak sounds. It is for this reason that there must be more than one frequency-weighting network.

(3) The potentiometer varies the input to the amplifier. It is calibrated in 5-*db* steps to provide various scales for measuring a wide range of noise levels.

(4) The amplifier increases the very small amplitude of the noise signal, so that it may cause a deflection of the meter needle.

(5) The copper-oxide rectifier converts the a-c noise signal into a d-c signal so that it can be measured on a more sensitive d-c ammeter. The noise signals are combined by the rectifier to produce an average d-c signal in the manner by which the ear combines noise signals.

(6) The noise meter is a d-c ammeter calibrated in decibels.

b. On the front panel of a noise-measuring set are the input terminals; the calibrated potentiometer, with 5-*db* steps from 15 *db* to 75 *db*, to provide various scales; the calibrate-adjust screwdriver control which varies the current in the meter; the meter; and the weighting-circuit selector switch with three positions (fig. 241). The headphone provides a means of hearing the noise as it is being measured.

c. The noise-measuring set can be used to make measurements directly on a transmission line. In combination with another transmission-measuring set, the noise-measuring set is used for frequency and amplitude analysis of the noise signals induced on telephone circuits, and for measurements of harmonic signals on power circuits. An accurate harmonic analysis of the noise signals on a telephone circuit often indicates the source of the interference on the line. When the set is used for measuring noise, the circuit to be measured is connected to binding posts on the front panel of the measuring set. The noise level is then read on the calibrated potentiometer and the meter. For example, if the potentiometer is set at the +35-*db* level and the meter reads +2.5 *db*, the noise level is +37.5 *db RN*. The measuring set has been calibrated at 1,000 cycles to read 0 *db RN* for an input signal of 1 micromicrowatt.

d. Calibration of the noise-measuring set requires an external oscillator. The oscillator is adjusted for a frequency of 1,000 cycles. The control resistance in the oscillator is adjusted until the meter reads 10 microamperes. This calibrating current flows in the 600-ohm input circuit of

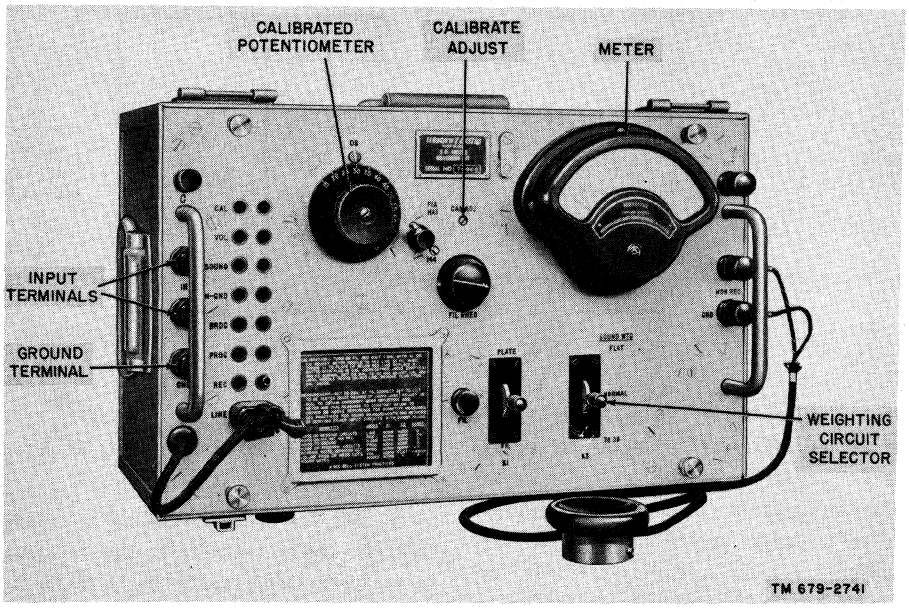


Figure 241. Noise-measuring set.

the noise-measuring set. The power in the 600-ohm resistance is

$$P=I^2R$$

$$=(10\times10^{-6})^2(600)$$

$$P=6\times10^{-8} \text{ watt.}$$

The power level above 0 *db RN* (1 micromicrowatt$=10^{-12}$ watt in the 600-ohm resistance) is

$$N_{db}RN=10 \log \frac{6\times10^{-8}}{10^{-12}}$$

$$=10 \log 6\times10^4$$

$$=47.8 \text{ } db \text{ } RN.$$

The dial for the calibrated potentiometer now is set at +45 *db*, and the calibrate-adjust screwdriver control on the front panel is adjusted until the ouptut meter reads +2.8 *db*. The total noise-level reading is 47.8 *db* above 0 *db RN*. Setting up the 0 level itself on the output meter is a difficult operation, because the current required in the 600-ohm resistance is almost too small to measure (.04 microampere). The calibration can be checked by noting the accuracy of the reading on the noise-measuring set for some other current in the 600-ohm resistance. For a current of 8 microamperes, the meter and potentiometer should read +45.8 *db RN*.

210. Summary of Methods of Transmission Measurement

a. Two types of measuring sets can be used to measure the power losses and gains in a network. The first is a comparison-type measuring set.

(1) A 1,000-cycle test tone is applied to an artificial line (calibrate), which is calibrated to cause a fixed amount of attenuation. The test signal amplitude is adjusted so that the meter reads midscale.

(2) The test tone is switched over to a variable artificial line (measure), and the attenuation is varied so that once again the meter reads midscale. This variable line now causes the same amount of attenuation as the calibrated line.

(3) The network being measured is connected in series with the variable line. The total attenuation caused by the variable line and the network is greater than the attenuation resulting from the variable line alone. The attenuation caused by the variable line is reduced so that the meter again reads midscale. The total attenuation due to the variable line and the network is equal to the attenuation due to the calibrated line. The attenuation being measured is equal to the difference between the attenuation reading on

the calibrated-line dial and the attenuation reading on the variable-line dial.

(4) The measuring set can be used also to measure circuit gains. In this case, the attenuation caused by the variable line is increased so that the meter reads mid-scale. The gain being measured is the difference between the attenuation readings on the calibrated and variable line dials.

b. The second type of measuring set reads network losses and gains directly.

(1) A 1,000-cycle, 1-milliwatt sine-wave test tone is applied to the network.

(2) The output of the network is read on a d-c meter calibrated in decibels. An output of 1 milliwatt causes a meter reading of 0 *dbm*. An output of 10 milliwatts causes a meter reading of +10 *dbm*. An output of 0.01 milliwatt causes a meter reading of −20 *dbm*.

(3) Key-controlled pads change the range of the measuring set.

(4) The sensitivity of the measuring set is increased by the addition of an amplifier. Negative feedback in the amplifier improves its stability. The range of the measuring set is changed in steps of 10 *dbm* by varying the resistance in the cathode circuit of the amplifier.

c. A volume indicator is a measuring set for measuring signals which contain components of many different frequencies (speech, music), as compared with the usual measuring set which measures only single-frequency signals (usually 1,000 cycles).

(1) The volume indicator is a voltage-operated device, calibrated to read speech-volume levels across a 600 ohm resistor. The reference power level is 1 milliwatt in the 600 ohm resistor. Speech-volume levels are indicated on a decibel scale.

(2) Speech volume levels are expressed in terms of a unit called the *vu*. The *db* is associated with steady, single-frequency signals. The *vu* is associated with signals containing many components of different frequency and amplitude.

(3) The meter would read 100, or 0 *vu*, on a 0-*vu* scale, when the power in the 600 ohm resistor is 1 milliwatt. There is no such scale on the *vu* meter, however. On the lowest scale, which is a +4-*vu* scale, the meter reads 0 *vu* when the power in the 600-ohm resistor is 4 *db* above 1 milli watt—that is, 2.512 milliwatts.

(4) When the volume indicator is set on the +4-*vu* scale, a reading of 0 *vu* on the meter indicates that the speech-volume level is 4 *db* above 1 milliwatt of speech volume. The true volume level is the algebraic sum of the meter reading and the scale value.

d. Crosstalk exists when the signal being transmitted on one communication circuit (called the disturbing circuit) can be detected in another communication circuit (called the disturbed circuit).

(1) Crosstalk coupling is the ratio of the transmitted power in the disturbing circuit and the induced power in the dis turbed circuit. Such coupling is expressed in decibels.

(2) In a crosstalk-measuring set, an oscillator supplies a test signal to the disturbing circuit and to a calibrated attenuator. The test tone appears in the disturbed circuit because of crosstalk coupling. The output of the calibrated attenuator also appears in the disturbed circuit.

(3) A switch is thrown from the disturbed circuit to the calibrated attenuator, causing the test signal to be applied to both circuits. The attenuation is varied until the signal heard in the receiver (or read on a meter) is the same on both positions of the switch. The attenuator is calibrated in decibels. The crosstalk coupling is read directly on the attenuator dial.

e. Noise is a collection of sounds that has a disturbing effect on the ear. Noise can appear in a communication circuit as a result of coupling between the circuit and some other communication circuit or a powerline circuit. Voltages and currents, other than those caused by the transmitted signal appear as noise in the receiver at the end of the line.

(1) The disturbing effect of noise depends on its volume and on the frequency of the noise signal. Noise signals having a frequency in the 800-to-1,200-cycle range are far more disturbing than those in the higher and lower ranges.

(2) When measuring the noise in a communication circuit, the noise meter is cali-

brated to read 0 *db RN* for a reference power of 1 micromicrowatt at a frequency of 1,000 cycles. Noise levels then are indicated in terms of N_{db} above or below the reference level.

(3) The noise-measuring set is a direct-reading measuring set. The noise is read on a calibrated potentiometer and on a meter; the total noise level is the sum of the potentiometer and meter readings. When the potentiometer is set at the 35-*db* level and the meter reads +2.5 *db*, the noise level is +37.5 *db RN*.

(4) The noise-measuring set is calibrated with an external oscillator. A 1,000-cycle signal is supplied by the oscillator. This signal develops 6×10^{-8} watts of power in the 600-ohm input circuit of the measuring set. This is 47.8 *db RN* above 0 *db RN*. The calibrated potentiometer is set at +45 *db*, and the current in the meter is adjusted until the meter reads +2.8 *db*.

211. Review Questions

a. What power ratio is expressed by 1 bel? By 1 decibel?

b. Define the decibel. What is the abbreviation for the decibel? What is the abbreviation for a number of decibels?

c. Is the decibel a unit of amount of power change? Is it a unit of power ratio?

d. In the formula $N_{db}=10 \log \frac{P_1}{P_2}$, define N_{db}, P_1, P_2. When is a N_{db} positive? When is it negative? When is P_1 the input power? When is it the output power?

e. Determine the number of decibels of power change in the following examples, stating in each case whether there has been a power gain or power loss.

Examples:

(1) Output power=100 watts; input power=1 watt.

(2) Output power=5 milliwatts; input power=2.5 milliwatts.

(3) Input power=7.2 milliwatts; output power=0.8 milliwatt.

(4) Output power=20 kilowatts; input power=80 kilowatts.

Answers:

(1) $N_{db}=+20$ (gain).

(2) $N_{db}=+3$ (gain).

(3) $N_{db}=-9.54$ (loss).

(4) $N_{db}=-6$ (loss).

f. Determine the ratio of the output and input powers in the following examples, stating in each case whether there has been a power gain or power loss:

Examples:

(1) $N_{db}=+10$ *db*.

(2) $N_{db}=-30$ *db*.

(3) $N_{db}=-27$ *db*.

(4) $N_{db}=+150$ *db*.

Answers:

(1) $P_2/P_1=10$ (gain).

(2) $P_2/P_1=0.001$ (loss).

(3) $P_2/P_1=2/1{,}000=0.002$ (loss).

(4) $P_2/P_1=10^{15}$ (gain).

g. Assume that the four power changes in the preceding example exist along a telephone line. What is the total power change in the network?

Answer: 103 *db*.

h. How does the *dbm* differ from the *db*? Why is it useful to use a reference power level? Why is 1 milliwatt used as the reference power level? Why is a frequency of 1,000 cycles used for a test-tone signal?

i. Determine the output power of the networks in the following examples, in which the networks cause the power to change by the specified amounts. In each case, state whether there has been a power gain or power loss.

Examples:

(1) $N_{dbm}=+20$ *dbm*.

(2) $N_{dbm}=-40$ *dbm*.

(3) $N_{dbm}=-29$ *dbm*.

(4) $N_{dbm}=+.5$ *db*.

Answers:

(1) $P_o=100$ milliwatts (gain).

(2) $P_o=0.0001$ milliwatt (loss).

(3) $P_o=0.00125$ milliwatt (loss).

(4) $P_o=1.12$ milliwatt (gain).

j. For $N_{db}=db$, use the table of decibels to determine:

(1) Power-ratio factor for losses.

(2) Power-ratio factor for gains.

(3) Percent power loss.

(4) Percent power gain.

Answers:

(1) Loss power-ratio factor=0.016.

(2) Gain power-ratio factor=60.

(3) Percent power lost=98.4 percent.

(4) Percent power gain=5,900 percent.

k. Define: insertion loss, reflection loss, bridging loss.

l. Define the 0-level point. In a telephone network, what is the power level at the 0-level point? What piece of equipment in a telephone network is used as the 0-level point?

m. Draw a power-level diagram for the telephone network of figure 213.

n. What is the underlying principle of a comparison-type measuring set? The oscillator? Switch? Artificial (calibrate) transmission line? Variable (measure) artificial line? Variable impedance networks?

o. List, in proper order, the steps taken in operating a comparison-type measuring set. Explain the purpose of each step. What dial readings give the loss or gain being measured?

p. How does the measuring of gain differ from the measuring of loss on a comparison-type measuring set?

q. What is the underlying principle of a direct-reading measuring set? What is the purpose of each of the following components of the direct-reading measuring set: Oscillator? Filter? Copper-oxide rectifiers? D-c ammeter? Key-controlled pads? Calibrating resistance? Amplifiers? Variable cathode resistor in the amplifier?

r. List the steps taken in operating a direct-reading measuring set. Explain the purpose of each step.

s. Why does a d-c ammeter, calibrated to read decibels directly, have a non-linear scale? How can the d-c ammeter be made to have a linear *db* scale?

t. What are the advantages and disadvantages of a comparison-type measuring set? What are the advantages and disadvantages of a direct-reading measuring set? Which is most used?

u. How does a speech signal differ from a sine-wave signal? Why cannot a meter calibrated for sine waves be used for measuring speech signals? What special characteristics does the meter in a volume indicator have?

v. Why was the *vu* adopted as a unit of speech-power levels? How is the *vu* similar to the *db*? How does the *vu* differ from the *db*? From the *dbm*?

w. List the steps in the operation of a volume indicator. How is the meter read? If the meter is on the 10-*vu* scale and the meter reads −4 *vu*, what is the speech-volume level?

x. Briefly explain the theory of a volume indicator. Why is the input impedance of the meter high, compared with 600 ohms? Why is the meter calibrated across 600 ohms?

y. Define crosstalk, crosstalk coupling, disturbing circuit, distrubed circuit. What is the unit of crosstalk coupling? Define crosstalk loss. What is the unit of crosstalk loss?

z. Define near-end crosstalk, far-end crosstalk, signal-to-crosstalk ratio, crosstalk amplification, equal-level crosstalk.

aa. What is the purpose of each of the following components in a crosstalk measuring set: Warble oscillator? Calibrated shunt? Switch? Telephone receiver?

ab. List in proper order the steps taken when operating a crosstalk-measuring set. Where is the crosstalk measurement read?

ac. Define noise. What determines the disturbing effect of a noise? Define impulse noise, random noise, power-hum noise. Give an example of each. What are the sources of noise in a communication circuit?

ad. What is the unit in which noise measurements are expressed? What is the reference noise level? How is this level indicated?

ae. What is the purpose of each of the following components in a noise-measuring set: Impedance-matching circuit? Frequency-weighting circuit? Calibrated potentiometer? Amplifier? Copper-oxide rectifier?

af. List the steps taken in operating a noise-measuring set.

ag. List the steps taken in calibrating a noise-measuring set.

282463 O—54——16

APPENDIX I
LOGARITHMS AND DECIBELS

1. Purpose of Decibel Measurement

The measurement of the transmission characteristics of electrical communication systems and their components is based on a unit called the decibel. The decibel is one tenth of a bel, which is defined as the logarithm of a power ratio of the base 10. Its use has two advantages—the attenuation or gain of the components of a system may be added or subtracted by simple arithmetic to obtain a numerical expression of the overall attenuation or gain, and, as a logarithmic unit, it most closely expresses the response of the human ear to changes in sound intensity. A complete understanding of the decibel and the associated system of measurement requires a thorough grasp of some basic mathematics—in particular, the common system of logarithms, which is reviewed in this appendix.

2. Powers of 10

The common system of logarithms is based on powers of the number 10. This system is used widely in engineering as a concise method of expressing very large or very small numbers. Any positive, real number can be expressed as a power of 10. As a start, consider the following:

$$10\times10=100=10^2$$

$$10\times10\times10\times10=10{,}000=10^4$$

$$10\times10\times10\times10\times10\times10\times10=10{,}000{,}000=10^7.$$

The number 10, used two times as a factor, equals the number 100, or 10 to the second power (squared); 10 used four times equals the number 10,000 or 10 to the fourth power; 10 used seven times equals the number 10,000,000 or 10 to the seventh power. Note that, in the final expression of each case, the value of the superscript—called the exponent of the number 10, or the power to which the number 10 has been raised—is equal to the number of times 10 is a factor of the number, and also is equal to the number of zeros following the initial digit 1. Obviously, this notation—10 to an indicated power—is a convenient shorthand for expressing large numbers containing many zeros.

3. Operations with Powers of 10

a. Multiplication. The same shorthand is useful in multiplying large numbers containing many zeros. Consider the following:

$$10\times100\times1{,}000=1{,}000{,}000=10^1\times10^2\times10^3=10^6$$

$$1{,}000\times1{,}000{,}000=1{,}000{,}000{,}000=10^3\times10^6=10^9.$$

Note that in the final expression of each case the value of the exponent is equal to the sum of the exponents of the numbers multiplied, expressed as powers of 10. Now consider the following:

$$100^3=(10^2)^3=10^2\times10^2\times10^2=10^{2+2+2}=10^6.$$

Here (and always) the parentheses of such an expression as $(10^2)^3$ can be removed and the expression brought to a power of 10 by multiplying the exponents.

b. Division. A similar technique is useful in dividing large numbers containing many zeros. Consider the following:

$$\frac{10{,}000}{100}=\frac{10^4}{10^2}=10^{4-2}=10^2.$$

Here (and always) subtracting the exponent in the denominator from the exponent in the numerator gives the correct result as a power of 10. *Note*:

$$\frac{10}{10}=\frac{10^1}{10^1}=10^{1-1}=10^0=1.$$

Thus, the number 10 raised to the zero power is equal to unity.

c. Roots. The root of a number expressed as a power of 10 can be found by similar techniques. Consider the following:

$$\sqrt{100}=\sqrt{10^2}=10^{\frac{2}{2}}=10^1=10$$

$$\sqrt[4]{100,000,000}=\sqrt[4]{10^8}=10^{\frac{8}{4}}=10^2=100.$$

In such expressions as $\sqrt{10^2}$ and $\sqrt[4]{10^8}$, then, dividing the exponent under the radical by the root of the radical (2 and 4 in the cases above) gives the correct result as a power of 10.

d. Negative Exponents. The number 10 raised to a negative exponent equals 1 divided by 10 raised to the same exponent but with a positive sign. Thus,

$$\frac{1}{10^{-1}}=10^1,\ \frac{1}{10^{-2}}=10^2.$$

Here, changing 10 to a power from the denominator to the numerator merely involves changing the sign of the exponent. Other examples of switching from denominator to numerator or vice versa are:

$$\frac{1}{10^{-2}}=10^2,\ \frac{1}{10^2}=10^{-2},\ 10^{-3}=\frac{1}{10^3},\ 10^3=\frac{1}{10^{-3}}.$$

Note that the sign of the exponent always changes. By using negative exponents we have

$$1/1000=10^{-3},\quad 1/100=10^2,\quad 1/10=10^{-1},\quad 1=10^0,$$
$$10=10^1,\quad 100=10^2,\quad 1000=10^3,$$

and so on.

e. Numbers Not Powers of 10. The expression of numbers as powers of 10 is a convenient shorthand. But the shorthand can be used for numbers that are not powers of ten. For example,

$$200,000=2\times 100,000=2\times 10^5$$
$$1,200,000=1.2\times 10^6.$$

Multiplication: $2,000\times 6,000=2\times 10^3\times 6\times 10^3=12\times 10^6.$

Division: $\frac{2,000}{60,000}=\frac{2\times 10^3}{6\times 10^4}=\frac{2}{6}\times 10^{-1}=\frac{1}{3}\times 10^{-1}=\frac{1}{30}.$

f. Any Numbers. These techniques are not restricted to powers of 10. They can be used with any number that is raised to a power. Thus, $2^5\times 2^3=2^8$, and $3^5\times 3^{-2}=3^3$.

4. Uses of Powers of 10

Consider an example of what can be done with the techniques using powers of 10.

a. Find the number represented by 246^{35}. Rewriting the expression gives 246^7. The answer, then, will be the square root of the quantity $246\times 246\times 246\times 246\times 246\times 246\times 246$. This method would require a tremendous amount of work. But, from the table in *5b*, this appendix, it is found that 246 expressed as a power of 10 is

$$10^{2.3909}$$

and, therefore, $246^{3.5}=(10^{2.3909})^{3.5}=10^{2.3909}\times 3.5=10^{8.3631}$. The same table, by a reverse procedure, is used to find that number which is 10 raised to the power 8.3681. The number is 233,390,000.

b. In the problem in *a* above, the use of methods associated with the expression of numbers as powers of 10 drastically reduced the necessary work. There are other problems, almost impossible to solve by arithmetic, but easily solved by the same methods. Such a problem would be the finding of the seventh root of any number; for example, $\sqrt[7]{3690}$. Obviously, it is highly advantageous to have a table in which most numbers are expressed as powers of 10.

5. Logarithms

The table in *b* below is called the table of logarithms.

a. To understand the meaning of this table, let $10^2=100$ be written in this form: $\log_{10} 100=2$. This means that the logarithm of the number 100 to the base 10 is equal to 2. It is another way of saying that the number 100 equals the base number 10 raised to the power 2. Again, $246=10^{2.3909}$; this expressed as a logarithm, is $\log_{10} 246=2.3909$. The logarithm of a number is the power to which the base 10 must be raised to produce the number. (Usually, the subscript "10," indicating the base, is omitted when all the logarithms dealt with are to the base 10 and there can be no confusion. Form now on it will be omitted.)

b. A portion of a table of logarithms follows:

	0	1	2	3	4	5	6	7	8	9
24	3802	3820	3838	3856	3874	3892	3909	3927	3945	3962
25	3979	3997	4014	4031	4048	4065	4082	4099	4116	4133
26	4150	4166	4183	4200	4216	4232	4249	4265	4281	4298

c. Logarithms consist of two parts as follows: that part in front of the decimal point, called the

characteristic, and all that follows, called the mantissa. The characteristic of the logarithm of any number that is unity or more than unity is always one less than the number of digits before the decimal point of the number. Thus, the characteristic of the logarithm of 246 is 2. Mantissas are listed in tables of logarithms; they are always decimals, but in the tables the initial decimal points, for simplicity, are omitted. In finding the mantissas of the logarithms of all numbers, the sequence of digits of the numbers is followed (except for initial zeros, as in 0.00246), and any decimal points in the numbers are ignored. Thus, in the table in *b* above, the mantissa of the logarithms of 246, 24.6, 2.46, 0.246, 0.0246, and 0.00246 is found in the line beginning with the number 24, under column 6; it is 0.3909. The logarithm of 246, then, is 2.3909; of 24.6 the logarithm is 1.3909, and of 2.46 it is 0.3909. To understand what has been done, note that the characteristic is the minimum exponent to which 10 must be raised to produce the number. Since, for 100, the exponent of 10 is 2, and, for 1,000 the exponent is 3, for every number between 100 and 1,000 (or having three digits) the exponent must be at least 2. For the number 246,

$$246=2.46\times10^{2}=10^{0.3909}\times10^{2}=10^{2+0.3909}=10^{2.3909}.$$

The characteristic, 2, takes care of the 100, and the mantissa, 0.3909, found in the table, takes care of the amount by which 100 has been multiplied to produce the number 246.

d. The characteristic of the logarithm of a number less than unity is negative. It is found by counting to the right the number of places required to move the decimal point to a position between the first two digits. The number of places so counted is the characteristic; to indicate that the characteristic is negative, a bar is placed over it. Thus, the logarithm of 0.246 is $\bar{1}.3909$; of 0.0000246 it is $\bar{5}.3909$. Note that only the characteristic is negative; the mantissa is positive. Thus $\bar{1}.3909$ means $-1+0.3909$, not -1.3909. Logarithms with a bar cannot be treated as ordinary numbers.

e. In transmission-line calculations it is necessary that the logarithms be written as ordinary numbers—either positive or negative—which can be added or subtracted. The logarithms of unity and numbers larger than unity are such numbers. As usually written, the logarithms of numbers less than unity are not ordinary numbers, but have a negative part and a positive part. However, they can be written as pure numbers by computation as follows:

$$0.0246=\frac{246}{10{,}000}$$

$$\log .0246=\log\frac{246}{10{,}000}=\log 246-\log 10{,}000$$

$$=2.3909-4.0000$$

$$=-1.6091.$$

A more practical method of finding a negative logarithm consists of first writing it with the bar, and then adding 10 to the characteristic while sub-subtracting 10 from the entire logarithm; thus,

$$\log 0.0246=\bar{2}.3909=8.3909-10.$$

One more operation converts the logarithm to an ordinary negative number. This is done by adding -10 to the entire logarithm, while adding $+10$ to its right-hand member, thus:

$$\begin{array}{r} -10.0000+10 \\ 8.3909-10 \\ \hline -1.6091 \end{array}$$

This result checks with the one found by the division method above.

f. The procedure of finding the number a logarithm represents is called finding its antilogarithm. It is the reverse of finding a logarithm. The mantissas in the table are scanned to find the one nearest in value to the one in the logarithm; the number at the left of the line in which it is found it written down, followed by the digit at the top of the column in which it is found. Thus, the antilogarithm of 0.4281 is found on line 26 under column 8, giving the series of digits 268. Since the characteristic is positive, the decimal point is put one place to the right (one more than the characteristic, which is zero). The antilogarithm, then, is 2.68. The antilogarithm of 2.4281 is 0.0268; since the characteristic is negative, the decimal point is put two places to the left of the position between the first two digits.

6. Units for Measurement of Transmission-Line Losses and Gains

a. Now that the mathematical foundation has been established, it is possible to proceed with the development of a method for measuring and expressing transmission-line losses and gains. At one time, it was customary to express the losses in

terms of the loss caused by a "standard cable mile." This was the loss in 1 mile of a certain standard 19-gage cable having a distributed resistance of 88 ohms per loop mile, a distributed capacitance of 0.054 microfarad per loop mile, and negligible distributed inductance and conductance. The loss varied with frequency because of the distributed capacitance, causing greater attenuation at the higher voice frequencies. The attenuation increased by 12 percent as the frequency increased from 800 to 1,000 cycles, indicating that the percentage reduction in power caused by the insertion of a mile of standard cable was different at different frequencies. Obviously, it was meaningless to say that a line was the equivalent of a certain number of miles of standard cable unless the frequency was given.

b. It became desirable, therefore, to find a new unit that would be independent of frequency, and which (for practical purposes) would have nearly the same values as the old unit. If a single-frequency standard must be used, the frequency should be in the 800- to 1,000-cycle range, because the attenuation of a signal in this frequency range is most nearly like the attenuation of a voice signal. The attenuation of a signal of frequency 886 cycles on a mile of standard cable is such that

$$\frac{P_1}{P_2}=\frac{E_1I_1}{E_2I_2}=10^{.1}.$$

This ratio was adopted as a standard for all frequencies, and was called first the "transmission unit." Later, its name was changed to decibel, abbreviated *db*.

7. Decibels

a. The decibel is the unit finally chosen for expressing the amount of power loss in any type of network, such as a transmission line, and the amount of gain provided by an amplifier or repeater. The decibel is the unit of attenuation produced by a standard cable mile at a frequency of 886 cycles. This means, if P_1 (A, fig. 242) is the power at the input terminals of a standard cable mile and P_2 is the power at the output terminals, there has been produced *1 decibel of attenuation* when the input and output powers are in the ratio $10^{.1}$; that is,

$$\frac{P_1}{P_2}=10^{.1}=1.26.$$

b. Now, let a transmission line consist of 2 standard cable miles (B, fig. 242). On a uniform line, line sections of equal length cause the transmitted power to be attenuated in the same ratio (par. 16). This means that a line 2 standard cable miles in length causes the transmitted power to be attenuated twice in the ratio $10^{.1}$,

$$\frac{P_1}{P_2}\times\frac{P_2}{P_3}=10^{.1}\times10^{.1}=10^{.2}$$

$$\frac{P_1}{P_3}=10^{2(.1)},$$

and there are 2 decibels of attenuation.

c. Finally, let the line consist of N standard cable miles (C, fig. 242). Since the line is uniform —that is, all standard mile sections have uniform characteristics—each standard-mile length causes the transmitted power to be attenuated in the ratio $10^{0.1}$, and the power is attenuated in this ratio N times; therefore,

$$\frac{P_1}{P_2}\times\frac{P_2}{P_3}\times\ \ldots\ \times\frac{P_{N-1}}{P_N}=10^{.1}\times\ \ldots\ \times10^{.1}=10^{.N}$$

$$\frac{P_1}{P_N}=10^{N(.1)}.$$

There are N decibels of attenuation.

d. If only the input and output powers are known as the result of measurement, how can the total attenuation be expressed as a definite number of units of attenuation—in decibels? The logarithm of a number is the power to which the base 10 must be raised to produce the number. In the ration in *c* above, $N(.1)$ is the power to which 10 must be raised to produce the number P_1/P_N. This means that $N(.1)$ is the logarithm of P_1/P_N; that is, $N\times.1=\log P/P_N$. Therefore,

$$\log\frac{P_1}{P_N}=N(.1)=\frac{N}{10}$$

$$\frac{N}{10}=\log\frac{P_1}{P_N}.$$

Multiplying both sides of the equation by 10 gives

$$N_{db}=10\log\frac{P_1}{P_N}.$$

This result means that the total attenuation of the transmitted power is equal to N decibels of attenuation when the input power to the transmission line is P_1, the output power is P_N; N_{db} is found by the foregoing formula, N_{db} being a shorthand notation for number of decibels.

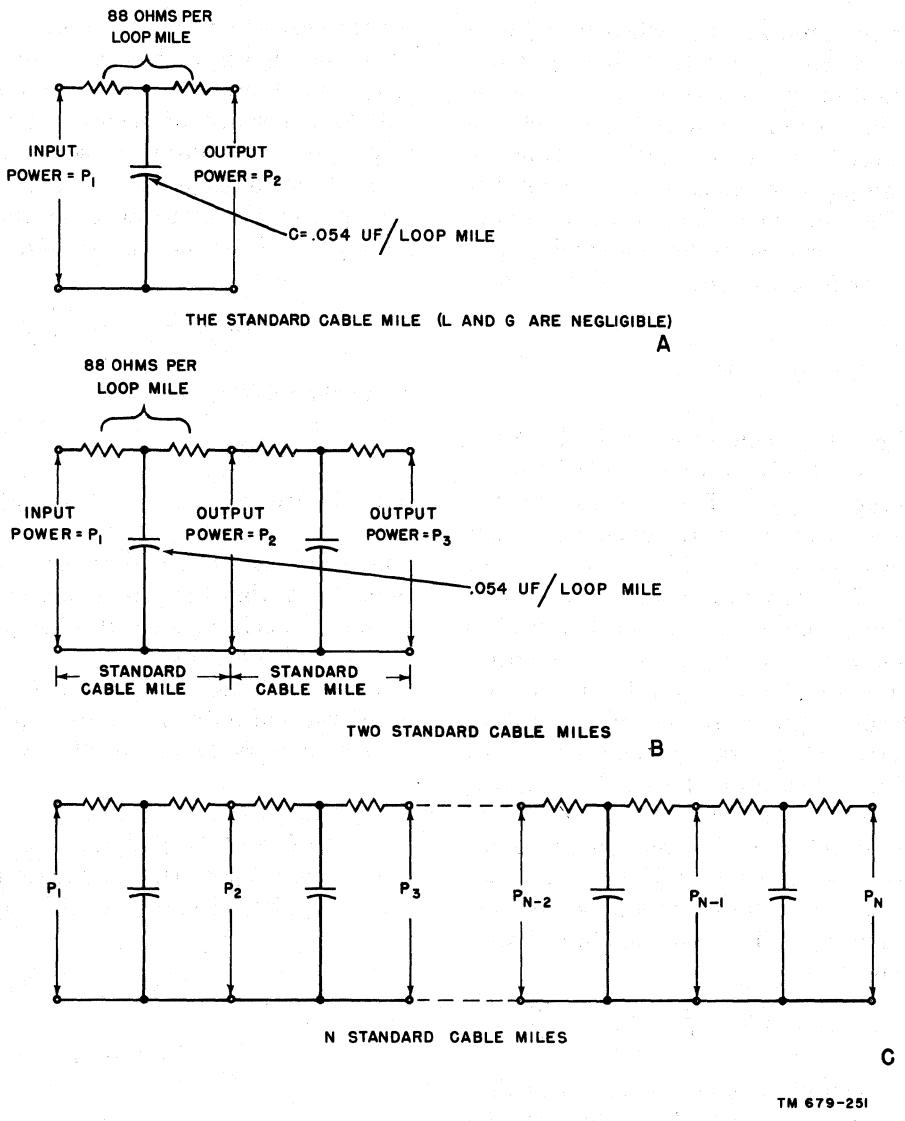


Figure 242. Standard cable mile.

It is the same as saying that the transmission line must be the equivalent of N standard cable miles, at a frequency of 886 cycles, so that there will be N decibels of attenuation, because every standard cable mile causes one decibel of attenuation.

e. It must be emphasized at this point that the decibel is fundamentally a unit of power ratio, not of power. When it is said that a transmission line causes 10 *db* of power attenuation, it means that the input and output powers are in a 10-to-1 ratio; that is, the output power is one-tenth of the input. But it does not state by what amount the power has actually changed. For instance, if the input power is 100 watts and the output power is 10 watts, the line bas caused 10 *db* of attenuation, and the amount of change is 90 watts. But if the input power is 1 watt and the output power is 0.1 watt, the line has again caused 10 *db* of attenuation but now the amount of change is 0.9 watt. The decibel system has given no indication of the difference in the amounts by which the power has changed.

8. Calculation of Decibel Loss or Gain

a. Decibels are also used to express the change of power caused by the insertion of equipment in the line. This power change may be either a decrease, or as in an amplifier, an increase. A method must be devised to distinguish between a number of decibels of power loss and a number of

decibels of power gain, so that, when it is stated that there have been N_{db} of power change, it will be known immediately whether the power change is a loss or gain. The method to be used should express all power losses as a negative number of decibels, and all power gains as a positive number of decibels. The following slight revision in the formula for N_{db}—a reversal of the input and output powers—gives the following formula, which affects this:

$$N_{db}=10 \log \frac{P_2}{P_1},$$

where P_2 is the output power and P_1 is the input power. Note that this is a different rule from the one in paragraph 184*a*. The purpose of the simplified procedure of 184*a* is to enable the reader to arrive at correct results with a minimum of mathematics. The formula given above, more mathematical in its approach, shows that if P_1 is always the input power and P_2 is always the output power, the number of decibels will take on its proper algebraic sign automatically. Either method, properly applied, leads to a correct result.

b. As an example of the use of decibels to express power losses, consider the power loss in a transmission line resulting from attenuation or the insertion of a piece of equipment into the line. Let the input power be 10 milliwatts and the output power 2.46 milliwatts. Then the number of decidebels of attenuation is

$$\text{attenuation } N_{db}=10 \log \frac{P_2}{P_1}=10 \log \frac{2.46}{10}.$$

Since the logarithm of a fraction is the logarithm of the numerator minus the logarithm of the nominator,

$$\log \frac{2.46}{10}=\log 2.46-\log 10.$$

Using the table of logarithms,

$$\log 2.46=.3909,$$
$$\log 10=1.0000.$$

therefore, $\log \frac{2.46}{10}=.3909-1.0000=-.6091$

and attenuation $N_{db}=10\times(-.6091)=-6.091$. The results shows that there have been 6.091 decibels of attenuation. Note that the attenuation is expressed as a negative number of decibels, the the minus sign having been introduced automatically by the mathematics of the method used.

c. As an example of the use of decibels to express power *gains*, consider the case of a repeater amplifier in a transmission line. Let the input power to the repeater be 2 milliwatts, and the output power 400 watts. Then the number of decibels of gain is

$$\text{gain } N_{db}=10 \log \frac{P_2}{P_1}=10 \log \frac{400}{0.002}=10 \log 200{,}000.$$

Using the table of logarithms,

$$\log 200{,}000=5.3010$$

and $$\text{gain } N_{db}=10\times(5.3010)=53.01.$$

The result shows that there are 53.01 decibels of gain. Note that the gain is expressed as a positive number of decibels, a result inherent in the mathematics of this method.

d. There remains the problem of finding the actual ratio of output and input powers when the ratio is expressed as a decibel figure.

(1) *Example 1*. When there is a power change of +12 *db*, find the ratio of output and input powers, and determine whether the power change is caused by loss or gain.

$$N_{db}=12=10 \log \frac{P_2}{P_1}$$

$$1.2=\log \frac{P_2}{P_1}.$$

From the table of logarithms it is found that the antilogarithm of the mantissa 0.2 is 1585. The characteristic is 1, indicating there must be two digits before the decimal place; therefore,

$$\frac{P_2}{P_1}=15.85.$$

The output and input powers are in the ratio 15.85, indicating that the output power is greater than the input (as already known, because the power change was expressed by a positive number of decibels). The change is one of gain.

(2) *Example 2.* When the power change is -25 *db*, find the ratio of output and input powers, and determine whether the power change is due to loss or gain.

$$-25=10 \log \frac{P_2}{P_1}.$$

Multiplying both sides by $-.1$.

$$2.5=-\log \frac{P_2}{P_1}=-(\log P_2-\log P_1)=\log P_1-\log P_2$$

$$2.5=\log \frac{P_1}{P_2}.$$

This result means that when the ratio of output and input powers is expressed as a negative N_{db}, the ratio of input and output powers can be expressed as a positive N_{db}. Concluding, from the table of logarithms it is found that 3162 has the logarithm mantissa 0.5. The characteristic is 2, indicating there must be three digits before the decimal place; therefore

$$\frac{P_1}{P_2}=316.2.$$

In this case, the input power P_1 and the output power P_2 are in the ratio 316.2, indicating that the input power is greater than the output power (as already known because the power change was expressed by a negative number of decibels). The change is one of loss.

INDEX

	Paragraph	Page
Alarm circuits	107	
Cable	110	
Carrier-supply	111	
Amplification:		
Need	38*b*	
Amplifier, electronic	2*a*, 54	
Application:		
44-type repeater	61	
Terminal repeater	62	
21-type repeater	57	
22-type repeater	58	
D-c amplifier	154*c*, 161	
Regulation of gain	109	
Single-stage	55	
Two-stage	56	
Use	38*c*	
With antihybrid devices	64*a*	
Amplitude modulation	4*b*(2)	
Use in carrier telegraph systems	151*a*	
Antihybrid devices	64	
Attenuation in transmission lines	16, 27*a*	
Causes	28	
Unit of measurement	183	
Automatic frequency control	175*b*(3)	
Automatic gain control	154*a*	
Bandwidth, carrier telegraph systems	169	
Balanced-bridge modulator	79, 80, 81	
Balancing network	44*a*, 52	
Bel	183	
Bias adjustment	141	
Bias distortion	131*d*, 132	
Bridging loss	186*c*	
Building block principle	5*g*	
Buildup time	169*a*	
Cable	25	
Alarm	110	
Coaxial	25*d*	
Crosstalk	34	
Lead-covered	25*a*	
Rubber-covered	25*b*	
Spiral-four	25*b*(2)	
Submarine	25*c*	
Capacitance:		
Distributed	32*a*(1)	
Shunt	9*d*	
Carrier:		
Definition	4*a*(2)	
Four-wire terminal	101, 102, 103	
Oscillator	4*b*(1)	
Purpose	1	
Suppression	4*b*(4)	
Telegraph terminal	121, 146, 147	
Terminals	100	
Two-wire terminal	104	
Carrier-frequency voltage switching:		
Balanced-bridge modulator	80	
Lattice-type modulator	83	
Carrier system:		
Basic elements	72	
Basic principle	4*b*	
CF series	5*a*	
Components	118	
C-type	5*d*	
Definition	4*a*(2)	
H-type	5*d*	
Integrated multichannel	115	
Military	5	
New four-channel	115	
Packaged	5*d*	
Purpose	71*a*	
Telephone terminal AN/TCC–3( )	5*f*, 115*b*	
Two-wire	105, 106	
Carrier telegraphy:		
Automatic gain control	154*a*	
A-m versus f-s systems	173*c*	
D-c amplifier	154*c*, 161	
Detector	154*b*	
Filters	154*b*	
Frequency drift	175	
F-s modulated using d-c loop	162	
F-s modulated using v-f, f-s loop	163	
Fundamentals	150	
Interaction with other services	172	
Interference	171, 174	
Level variations	176	
Maximum number of channels obtainable	170	
Modulator	153*b*	
Noise	173	
Receiving circuits	154	
Selection of carrier frequency	178	
Sending circuits	153	
Speech-plus-duplex	177	
Speed versus bandwidth considerations	169	
Transmission objectives	168	
Types of systems	151	
Frequency-shift	151*d*, 156, 157	
On-off method	151*b*, 155	
One-source two-band	151*f*	
Single-channel	151*a*	
Single sideband	151*c*	
Two-source	151*e*	
Two-source f-s receiving circuit	165	
Two-source f-s sending circuit	164	
Using on-off modulation and half-duplex loop	155	
Characteristic distortion	131*d*, 134	

	Paragraph	Page
Characteristic impedance:		
Definition	12	
Factors affecting	14	
Characteristics, line	9*a*	
Coaxial cable	25*d*	
Coils, repeating:		
Component of repeater	2*a*	
Conductance, shunt	9*e*	
Conductive coupling	32*d*	
Constants, distributed	9	
Control circuits	107	
Oscillator frequency	108	
Copper-oxide varistor	4*b*(6), 75, 78	
Electrical characteristics	76	
Crosstalk	30, 33*b*	
Coupling	202	
Measurement	201, 204	
Minimizing	34	
Types	203	
With repeaters	33*e*	
D-c amplifier	154*c*, 161	
Decibel	183*b*, app I	
Calculation of loss or gain	app I	
Correlation with *dbm* and *vu*	200	
Measurements	app I	
Meters	194	
Table	185	
Use	184, 186	
Demodulation	4*b*(6), 74, 150*b*(4)	
Demodulator	4*b*(6), 71*b*(3), 74, 78, 102*b*	
Detector, carrier telegraph system	154*b*	
Dielectric	9*d*	
Diode, germanium crystal	77, 78	
Discriminator	160	
Distortion:		
Bias	131*d*, 132	
Characteristic	131*d*, 134	
Fortuitous	131*d*, 135	
Distortion in transmission	20	
Frequency	20*b*	
Phase	20*c*	
Distributed constants	9	
Dotting speed	169*b*(3)	
Electromagnetic coupling	32*c*	
Electrostatic coupling	32*b*(1)	
Equalizers	96	
Component of repeater	2*a*	
Field wires	26	
Filter	4*b*(4), 85, 102*a*	
Band-pass	72*b*, 73*a*, 74*b*, 88	
Band-stop	88*e*	
Directional	104*b*	
High-pass	87	
In carrier telegraph system	153*c*	
Low-pass	72*d*, 86	
Fortuitous distortion	131*d*, 135	
Four-wire terminating set	47	
Frequency:		
Distortion	20*b*	
Division multiplexing	71*a*(2)	
Modulation	4*b*(3)	
Drift, effects	175	
Shift	4*b*(3)	
Carrier telegraph system	151*d*	
Elements of system	157	
Nature of signal	156	
System	157	
Amplifier limiter	159	
Modulator	158	
Oscillator	158	
Two-source receiving circuit	165	
Two-source sending circuit	164	
Using d-c loop	162	
Using v-f, f-s loop	163	
Versus a-m systems	173*c*	
Full metallic circuit	126	
Germanium-crystal diode	77, 78	
Ground return circuit	126	
Home copy	126*c*	
Hybrid arrangements	3, 102*a*	
Army types	68	
Balancing network	44*a*, 52	
Basic principles	40	
Coil	45	
Four-wire terminating set	47	
Impedance bridge	44	
Need	38*d*	
173 type repeating coil	46	
Repeating-coil	46	
Resistance-hybrid	51	
Transhybrid loss	50	
Transmission loss	49	
Use	2*b*(2), 53	
Ideal transmission line	19*a*	
Impedance:		
Bridge	44	
Characteristic	12, 14	
Input	12*b*	
Terminating	13	
Inductance per unit length of transmission line	9*c*	
Input impedance of transmission line	12*b*	
Insertion loss	186*a*	
Interference, carrier telegraph systems	171, 174	
Interference in transmission lines	27*b*, 30	
Crosstalk	30	
Electrical causes	32	
Noise	30	
Reduction	33	
Sources	31	
Lattice-type modulator	82, 83, 84	
Leakage	9*e*	

	Paragraph	Page
Level variations	176	
A-m systems	176*a*	
F-s systems	176*b*	
Limiters	97	
Directly connected	98	
Magnetically coupled	99	
Line characteristics	9*a*	
Line problems	27	
Attenuation	27*a*	
Interference	27*b*	
Loading of transmission line	29	
Effect	29*b*	
Function	29*a*	
Limitation	29*e*	
Loading coils	29*d*	
Methods	29*c*	
Logarithms	183*b*, app I	
Loop circuits	120, 139	
Direct-current	139*b*	
Neutral operation to negative battery full-duplex	142	
Neutral operation to negative battery half-duplex	140	
Neutral operation to positive battery half-duplex	143	
Two-path polar operation full-duplex	144	
Polarential operation half-duplex	145, 146	
Function	139	
Repeatered	120*b*	
Trunk	120*a*	
Unrepeatered	120*a*	
Voice-frequency	139*c*	
Loss:		
Bridging	186*c*	
Insertion	186*a*	
Mismatch	186*b*	
Reflection	186*b*	
Measurements:		
Comparison-type set	192	
Decibel meters	194	
Direct-reading set	193	
Noise	208, 209	
Relative transmission level	189*d*	
Set for transmission	191	
Speech	195	
Standard testing power	187	
Transmission-performance, need of	182	
Units of transmission	183, app I	
Volume indicator	196, 197, 198, 199	
Zero-level point	188	
Military multichannel carrier systems	5	
Mismatch loss	186*b*	
Modem	72*f*, 75	
Modulation	4*b*(2), 71*b*(1), 73, 150*b*(2)	
Amplitude	4*b*(2)	
Frequency	4*b*(3)	
Phase	4*b*(3)	
Modulator	4*b*(2), 71*b*(1), 102*a*, 72*a*, 78	
Balanced-bridge	79, 80, 81	
Carrier telegraph system	153*b*	
Frequency-shift system	158	
Lattice-type	82, 83, 84	
Neutral operation to negative battery half-duplex	140	
Neutral operation to negative battery full-duplex	142	
Neutral operation to positive battery	143	
Noise	30, 33*a*, 205, 206	
Effect on a-m carrier telegraph systems	173*a*	
Effect on f-s systems	173*b*	
Measuring set	209	
Open-wire lines	24, 33*d*	
Oscillator:		
Carrier	4*b*(1), 71*b*(1), 72*a*	
Frequency control	108	
Frequency-shift system	158	
Pads	89	
Basic attenuation theory	90	
Symmetrical H	93	
Symmetrical pi	92	
Symmetrical square	94	
Symmetrical T	91	
Phantom circuit	35	
Phase:		
Modulation	4*b*(3)	
Difference	17*c*	
Distortion	20*c*	
Shift in transmission line	17	
Shift constant	17*b*	
Velocity	17*a*	
Polarential operation (half-duplex)	145, 146	
Polar operation, advantages	133	
Polar relays	128	
Biasing current	132*c*	
Type 209–FG	165*e*	
Type 255	128*g*	
Power-level diagrams	189	
Power ratio	184	
Power transfer, maximum	15	
Radiotelephony	4*c*	
Rectifiers:		
Copper-oxide	75	
Germanium-crystal diode	77	
Reflection in transmission line	21	
Reflecting loss	186*b*	
Relative transmission level	189*d*	
Relays, polar	128	
Biasing current	132*c*	
Type 209–FG	165*e*	
Type 255	128*g*	
Repeater, telephone:		
Amplifiers	54	
Army types	68	
Carrier-frequency	112	

	Paragraph	Page
Repeater, telephone—Continued		
Crosstalk	33*e*	
Definition	2*a*	
44-type	2*b*(4), 61	
Four-wire system	39*b*, 112*c*	
Intermediate	39	
Purpose	1	
Spacing	63, 113	
Systems	39	
Terminal	39, 62	
Two-wire system	39*a*, 112*a*	
21-type	2*b*(4), 57	
22-type	2*b*(4), 58	
Repeating coil:		
Arrangement in two-wire repeater	46*d*	
Component of repeater	2*a*	
Construction	48	
Description	46*a*	
Four-winding	47*a*	
Four-wire terminating set	47	
Hybrid arrangement	46*c*	
173-type	46*a*	
Principle	46*b*	
Resistance per unit length of transmission line	9*b*	
Resistive unbalance	32*e*, 35*c*	
Ringing:		
Terminal-to-terminal	65	
20-cycle	66	
Voice-frequency	67	
Selector magnet	126*a*	
Series:		
Inductance	9*c*	
Resistance	9*b*	
Shunt:		
Capacitance	9*d*	
Conductance	9*e*	
Sidebands	4*b*(2)	
Singing	3*a*, 38*d*(1), 58*c*	
22-type repeater	59	
Singing-point tests	60	
Single-sideband transmission with suppressed carrier	73*b*	
Speech measurement	195	
Speech-plus-duplex systems	177	
Spillover	174*b*	
Spiral-four cable	25*b*	
Standard testing power	187	
Submarine cable	25*c*	
Telegraph systems, d-c:		
Neutral operation	125*b*	
Polar operation	125*c*, 133	
Theory	125	
Telegraph terminals, carrier	121	
Facilities	122	
Lines	122*a*	
Radio	122*d*	
Repeatered lines	122*b*	
Telephone carrier system	122*c*	

	Paragraph	Page
Telegraphy, carrier:		
Automatic gain control	154*a*	
A-m versus f-s systems	173c	
D-c amplifier	154*c*, 161	
Detector	154*b*	
Filters	154*b*	
Frequency drift	175	
F-s modulated using d-c loop	162	
F-s modulated using v-f, f-s loop	163	
Fundamentals	150	
Interaction with other services	172	
Interference	171, 174	
Level variations	176	
Maximum number of channels obtainable	170	
Modulator	153*b*	
Noise	173	
Receiving circuits	154	
Selection of carrier frequency	178	
Sending circuits	153	
Speech-plus-duplex	177	
Speed versus bandwidth considerations	169	
Transmission objectives	168	
Types of systems	151	
Frequency-shift	151*d*, 156, 157	
On-off method	151*b*, 155	
One-source two-band	151*f*	
Single-channel	151*a*	
Single sideband	151*c*	
Two-source	151*e*	
Two-source f-s receiving circuit	165	
Two-source f-s sending circuit	164	
Using on-off modulation and half-duplex loop	155	
Teletypewriter	118*a*	
Components of station	119	
Margin measurements	136	
Teletypewriter circuits:		
Full-duplex	126*d*	
Half-duplex	126*c*	
Neutral half-duplex	126	
Neutral system, waveshapes	127	
Teletypewriter systems:		
Polar, theoretical waveshapes	130	
Speech-plus-duplex	177	
Waveshape distortion	131	
Terminals, carrier:		
AN/TCC–3( )	115*b*	
Four-wire	101, 102, 103	
Telegraph	121	
Two-wire	104	
Transformer coupling	43	
Transmission line:		
Attenuation	16	
Cables	25	
Characteristic impedance	12, 14	
Classification	23	
Coaxial	25*d*	

	Paragraph	Page
Transmission line—Continued		
Conductive coupling	32*d*	
Distributed constants	9	
Distortion	20	
Electromagnetic coupling	32*c*	
Electrostatic coupling	32*b*(1)	
Equivalent network	10	
Field wires	26	
Frequency	20*b*	
Function	8	
Group velocity	19	
Ideal	19*a*	
Input impedance	12*b*	
Interference	27*b*, 30, 31, 33	
Line characteristics	9*a*	
Loading	29	
Maximum power transfer	15	
Mismatch loss	50*b*	
Open-wire	24, 33*d*	
Phantom	35	
Phase:		
Difference	17*c*	
Distortion	20*c*	
Shift	17	
Shift constant	17*b*, 18	
Velocity	17*a*	
Problems	27, 38*a*	
Radiation, from	11	
Reduction of interference	33	
Reflection	21	
Resistive unbalance	32*e*, 35*c*	
Submarine cable	25*c*	
Terminating impedance	13	
Transposition	33*f*	
Transmission line—Continued		
Unbalance:		
Between pairs	32*b*	
To ground	32*a*	
Transposition	33*f*	
Transmission-measuring set	191	
Comparison type	192	
Decibel meters	194	
Direct-reading type	193	
Two-path polar (full-duplex) operation	144	
Unbalance:		
Between pairs	32*b*	
Resistive	32*e*, 35*c*	
To ground	32*a*	
Varistors, copper-oxide	4*b*(6), 75, 78	
Electrical characteristics	76	
Velocity:		
Group	19	
Phase	17*a*	
Volume indicator	196	
Calibration	199	
Circuit	198	
Theory of operation	197	
Units	196*c*	
Use	199	
Volume unit	200	
Wheatstone bridge:		
Alternating-current	42	
Direct-current	41*a*	
Zero-level point	188	

[AG 353 (22 Oct 53)]

BY ORDER OF THE SECRETARY OF THE ARMY:

M. B. RIDGWAY,
General, United States Army,
Chief of Staff.

OFFICIAL:

WM. E. BERGIN,
Major General, United States Army,
The Adjutant General.

DISTRIBUTION:

Active Army:

Tech Svc (1); Tech Svc Bd (1); AFF Bd (ea Svc Test Sec) (1); AFF (5); AA Comd (1); OS Maj Comd (5); Base Comd (2); MDW (2); Log Comd (1); A (5); CHQ (2); FT (1); PMS & T 11 (1); Dep (1) except 11 (10); Lab 11 (2); SigC Field Maint Shops (2); Mil Dist (1); T/O & E's, 11–7(2); 11–15N(2).

NG: Same as Active Army except one copy to each unit.

Army Reserve: None.

For explanation of distribution formula, see SR 310–90–1.

☆ U. S. GOVERNMENT PRINTING OFFICE: O—1954